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CF No Answer

Call Forward PC2 to PC3 - CF No Answer

PC3 (RAS)
PC client
transferred to

PC1 (RAS)
PC originator

Served does the call forwarding
GK PC2 or GK

Setup

Facility (call rerouting)

Facility (call-rerouting return result)

Setup if served is not PC2 but some type of call server

Release Complete

Note 1: If the served is a Gatekeeper or Gateway then the Gatekeeper or Gateway must be aware of the call status on PC2.
Note 2: Make more sense to have the Gatekeeper as the served device since it receives the initial ARQ for call setup

Setup (divertingLegInfo2 invoke)

Alerting

Release Complete

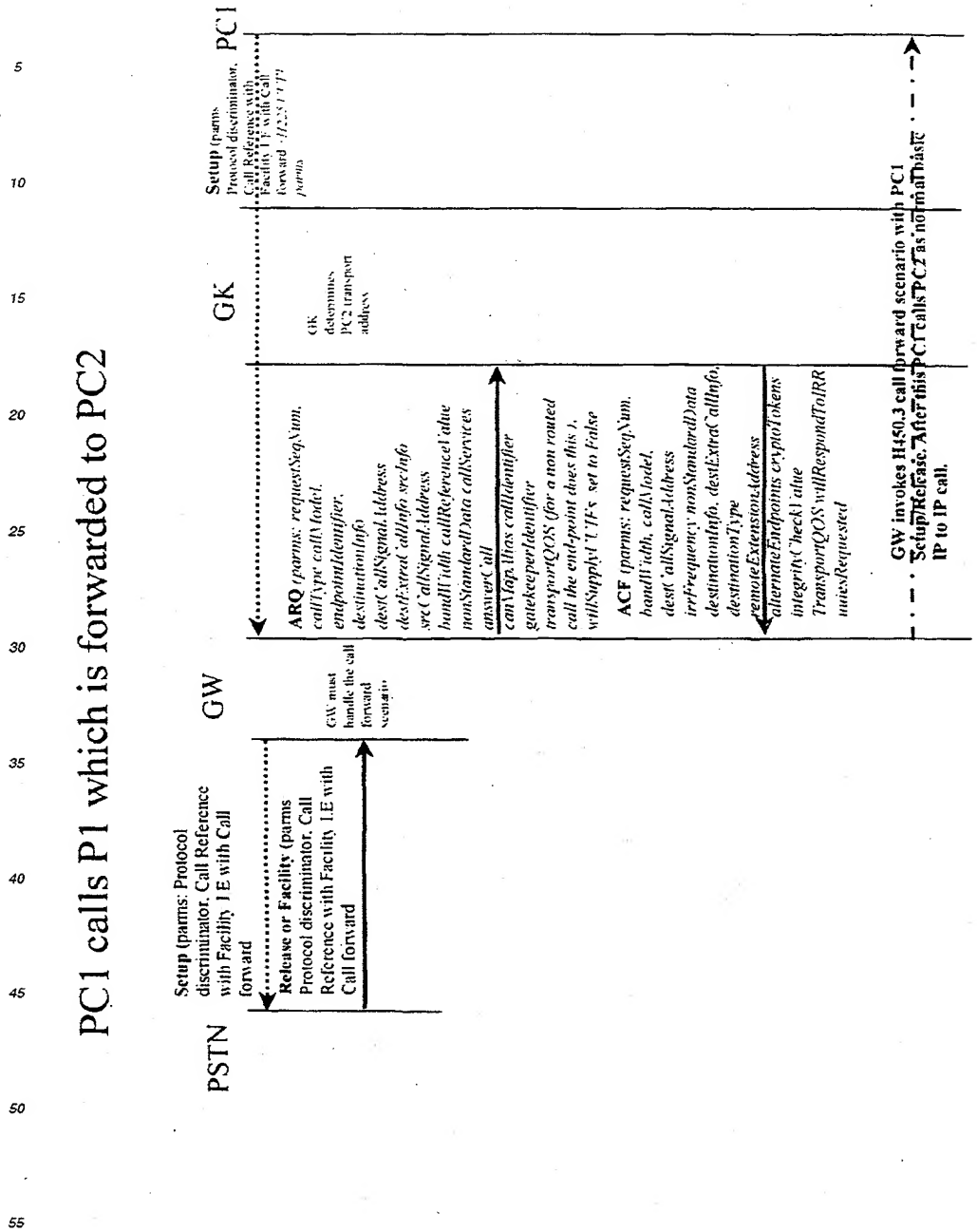
Connecting (divertingLegInfo3 invoke)

MEDIA channel

Call Forward Problems

If the originating terminal calls the PC1 (PC1 itself is responsible for call forwarding - SERVED). PC1 is registered but is not responding to setup messaging and hence will not forward the call. It is better to have the SERVED as the GK and possibly the Gateway. Since ARQ call queries are sent to the GK, it is logical to have the call forwarding functionality there also.

PC1 calls P1 which is forwarded to PC2



MLA & MADN

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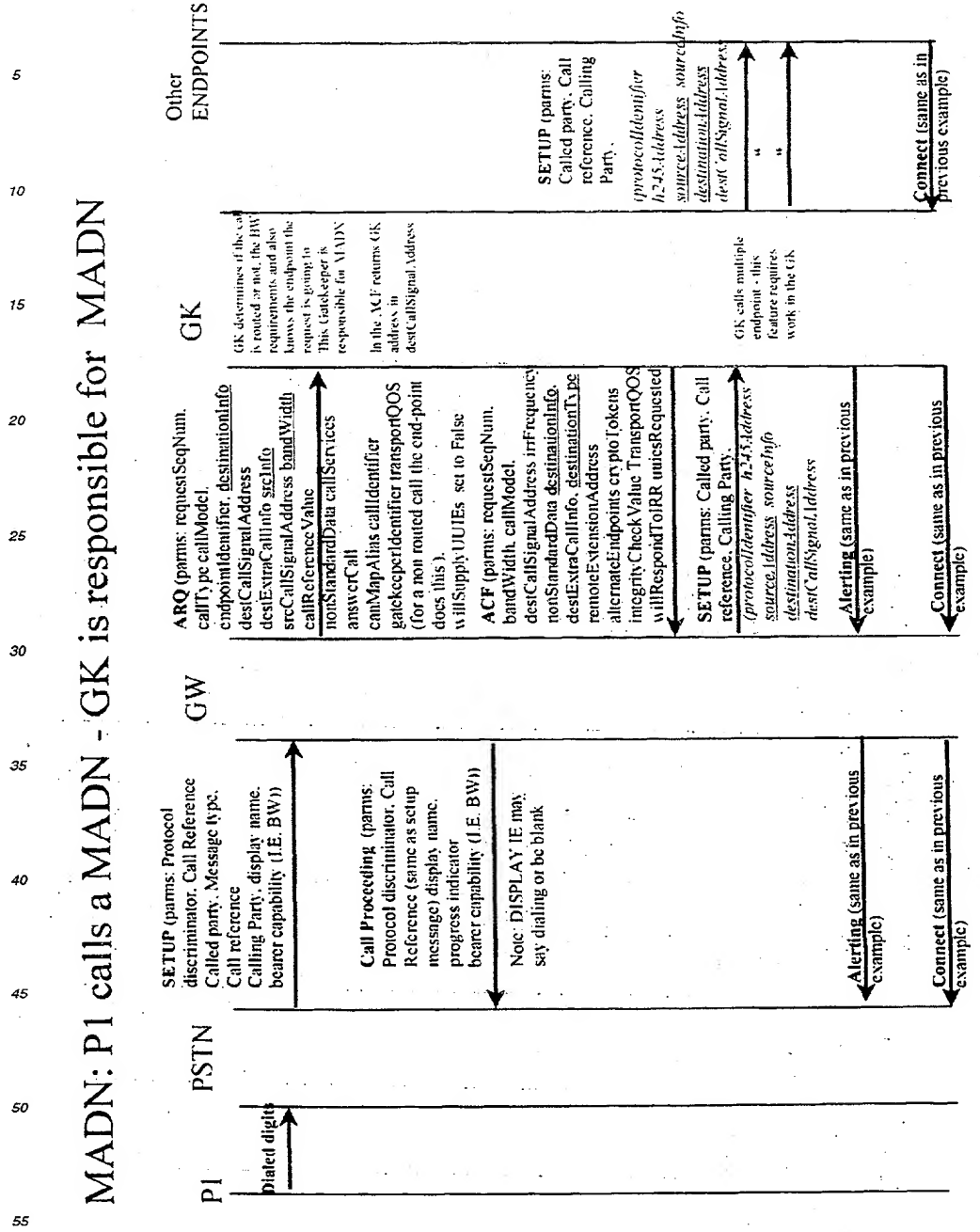
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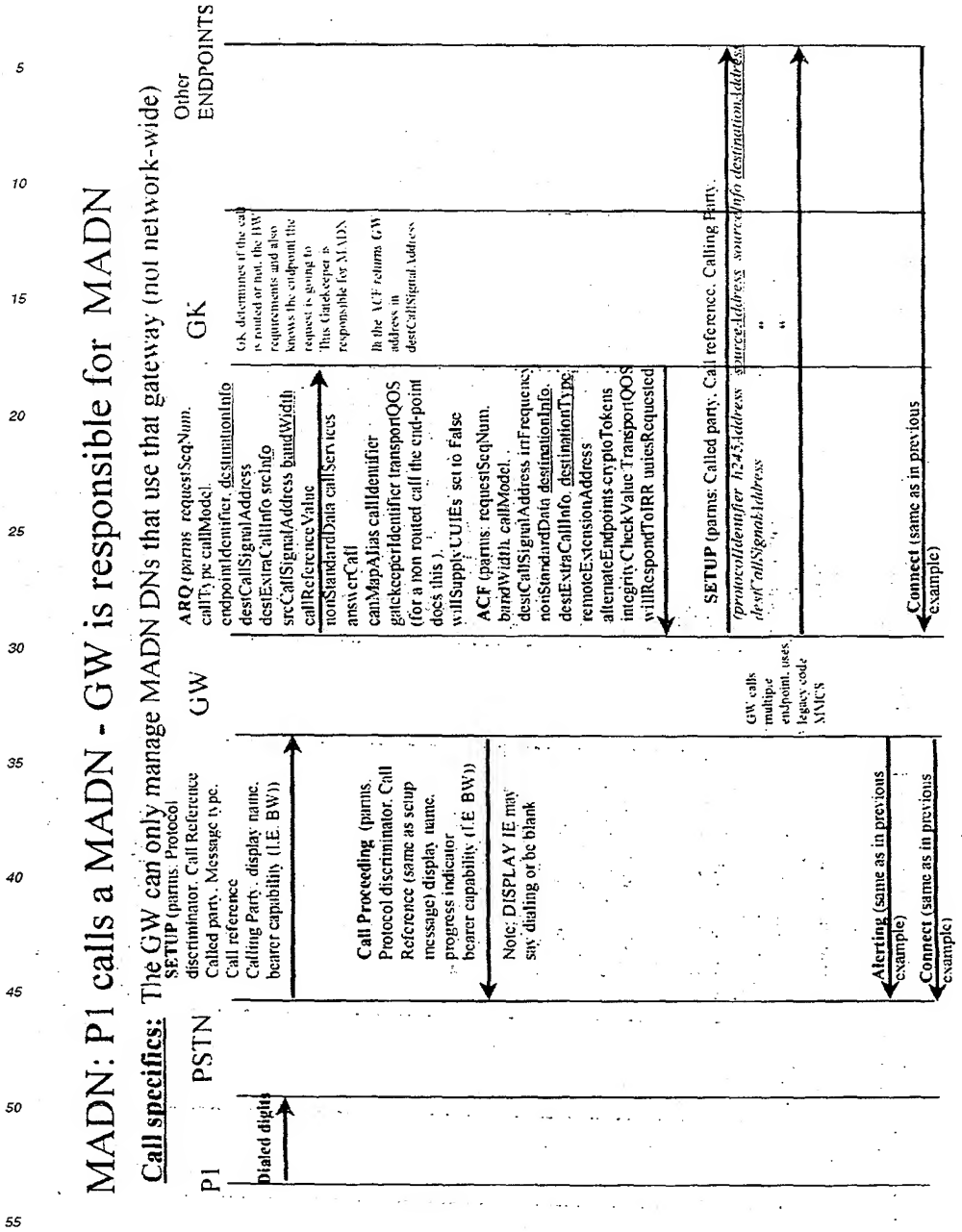
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MADN: P1 calls a MADN - GK is responsible for MADN



MADN: P1 calls a MADN - GW is responsible for MADN

Call specifics: The GW can only manage MADN DN's that use that gateway (not network-wide)



MLA: P1 calls a MLA PC1 - GK or GW handles

For MLA the Call Scenario is identical to the MADN scenarios for the GK and GW since these devices will handle the call setup. The media channel will be established after the call has been established and will be direct. The MMCS GW contains legacy code to do but will require modification, however for both the MADN and the MLA services managed by the Gateway, the features are restricted to those terminals served by this Gateway. The gatekeeper would need work for this feature to added

Both MADN and MLA do not require APDU supplementary services to be developed as these are features more capably handled by a Call Server device, I.E. GW or GK.

Voice Mail Call Flows

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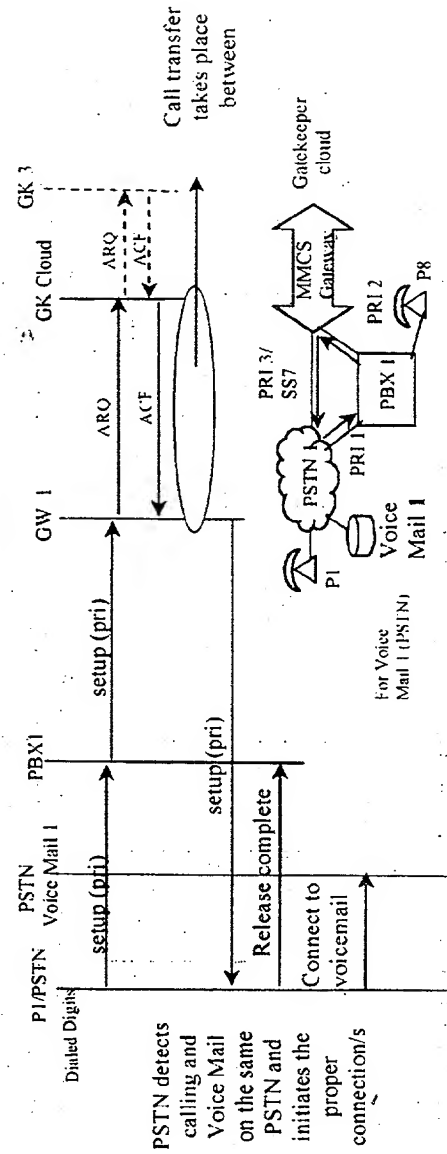
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P1 to P8 (voice mail on PSTN)

Call specifics.

Call from P1 to P8 (phone on PBX1). P8 is call forwarded to PC1 which is not registered. Voice Mail 1 on PSTN. Gatekeeper provisioned for WITH voice mail on the gatekeeper for P8. Gatekeeper uses H450.3 to route call to Voice Mail 1. This only applies for routed call scenarios.



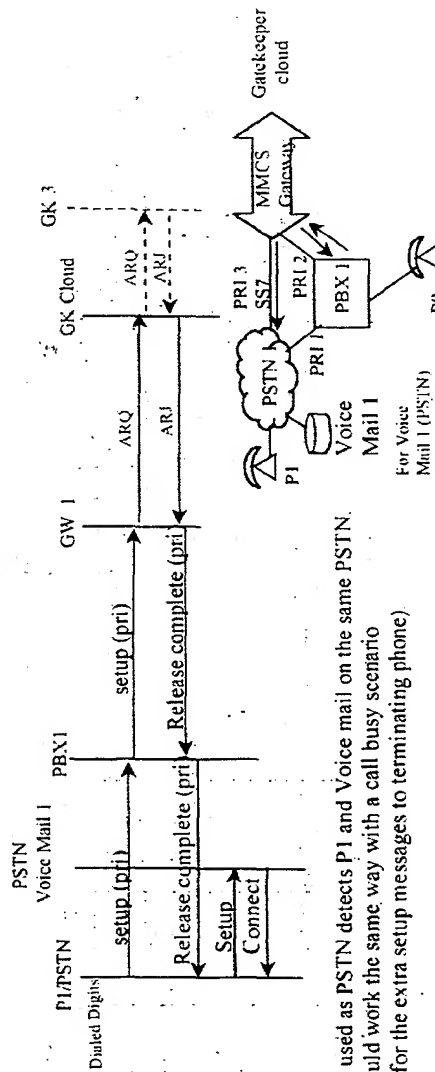
0 DSO's used as PSTN detects P1 and Voice mail on the same PSTN.

- Depending on the setup of the voice mail the callee may be required to enter the number of the phone of the called party, this is NOT desired functionality.

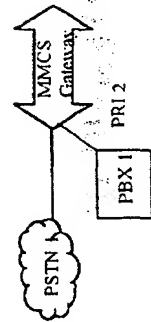
P1 to P8 (voice mail on PSTN)

Call specifics.

Call from P1 to P8 (phone on PBX1). P8 is call forwarded to **PC1 which is not registered**. Voice Mail 1 on PSTN. Gatekeeper rejects call. The PSTN knows that call cannot be terminated because of a release complete message, then the PSTN voice mail is to be used for P8.



0 DSO's used as PSTN detects P1 and Voice mail on the same PSTN. This would work the same way with a call busy scenario (except for the extra setup messages to terminating phone)

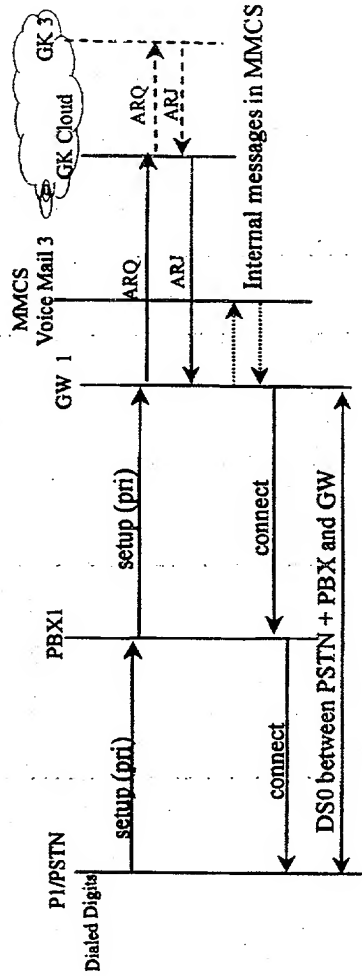


Another option is to have the PBX1 connected to MMCS directly. This would cause extra Q9:31 setup messages since all PBX messages will go through the MMCS. NOT GOOD !!

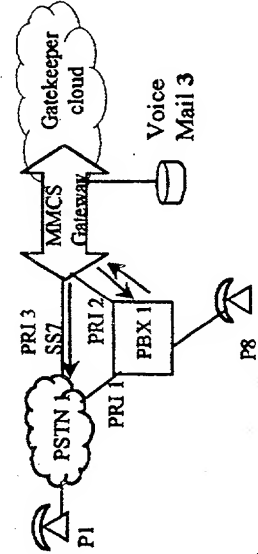
P1 to P8 (voice mail on MMCS/GW)

Call specifics.

Call from P1 to P8 (phone on PBX1). P8 is call forwarded to PC1 which is not registered. Voice Mail 3 on MMCS/GW.



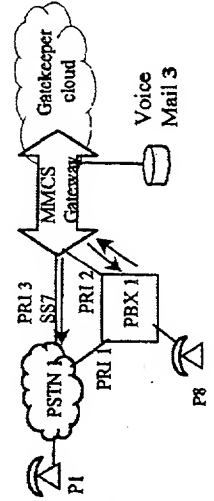
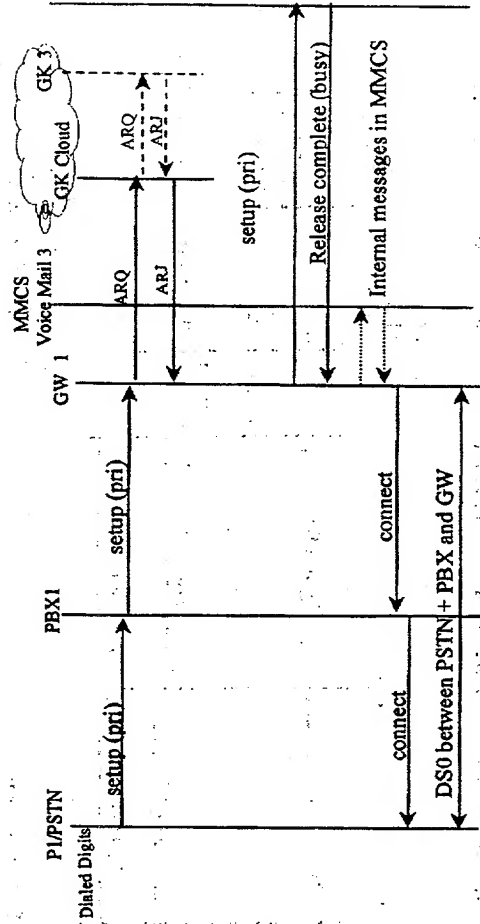
2 DS0's between PSTN and MMCS.



P1 to P8 (voice mail on MMCS/GW)

Call specifics.

Call from P1 to P8 (phone on PBX1). P8 is call forwarded to a BUSY FCI. Voice Mail 3 on MMCS/GW.

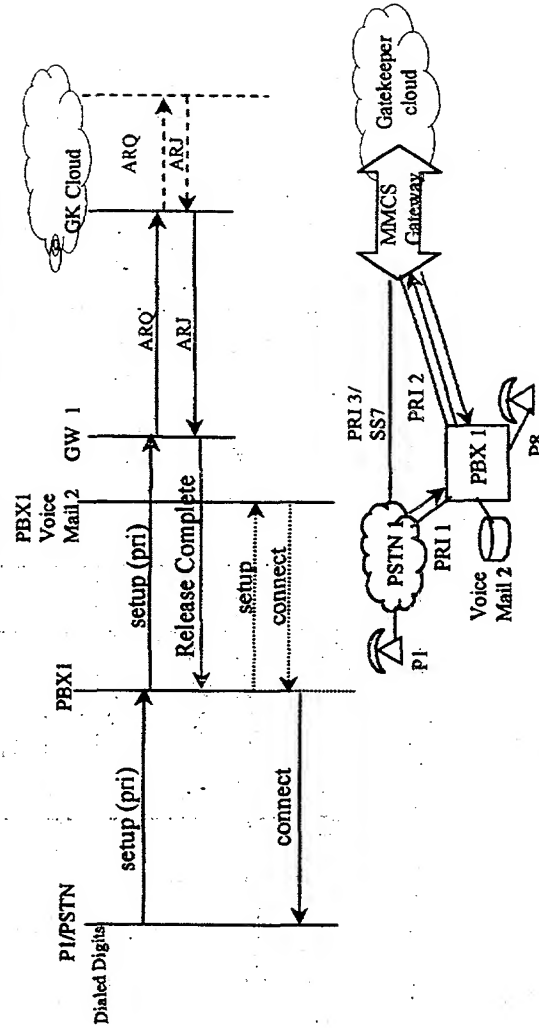


2 DS0's between PSTN and MMCS.

P1 to P8 (voice mail on PBX1 - express mail)

Call specifics.

Call from P1 to P8 (phone on PBX1). P8 is call forwarded to PC1 which is not registered. Voice Mail 2 on PBX1.



1 DS0 is taken by the call between P1 and Voice Mail 2.
Can the PBX handle a release complete and forward to a
internal mail? I Don't believe so!

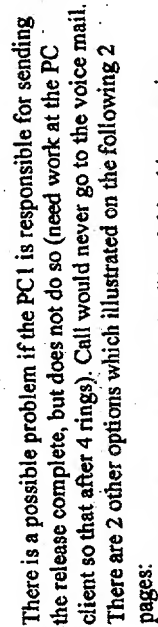
Call specifics.

Call Operator: Call from P1 to P8 (phone on PBX1). P8 is call forwarded to PC1 which is BUSY. Voice Mail 2 on PBX1.



Call specifics.

Call specifics. Call from P1 to P8 (phone on PBX1). P8 is call forwarded to PC1 but does not answer the phone. Voice Mail 2 on PBX1.

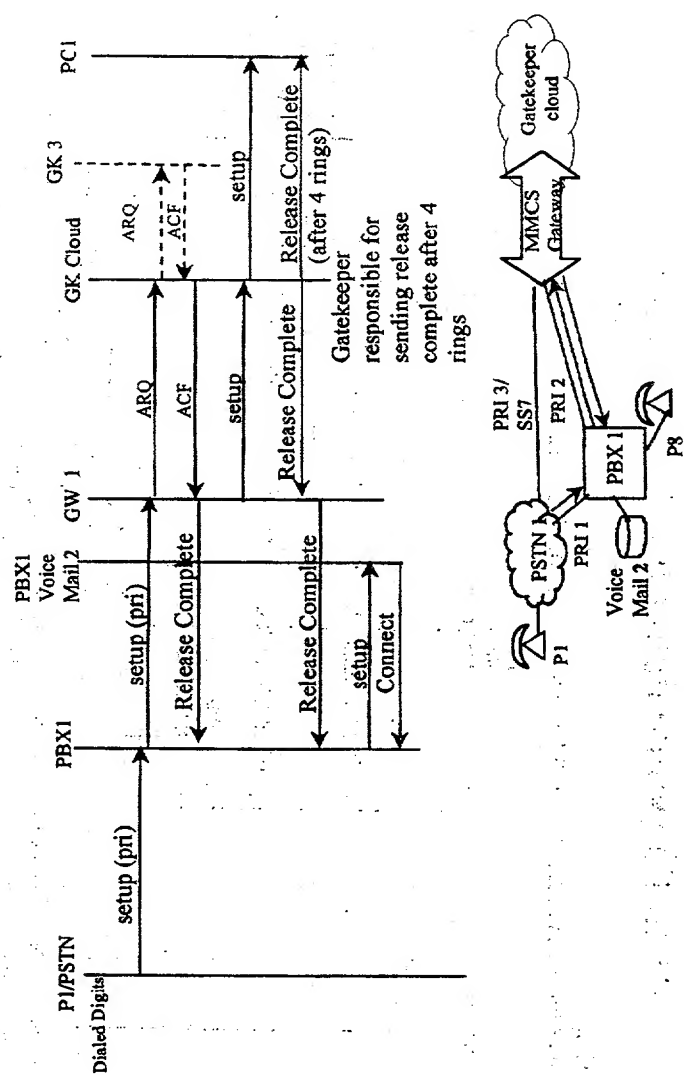


- It may be better to use a routed call model in this case via gatekeeper - Option 1
- After 4 rings the PBX sends a release complete to the gateway and connects to the PBX voice mail. Can the PBX do this presently? - Option 2

Call specifics.

Call from P1 to P8 (phone on PBX1). P8 is call forwarded to PC1 which is not answering. Voice Mail 2 on PBX1.

OPTION1: Gatekeeper handles call control (this only works for routed calls)

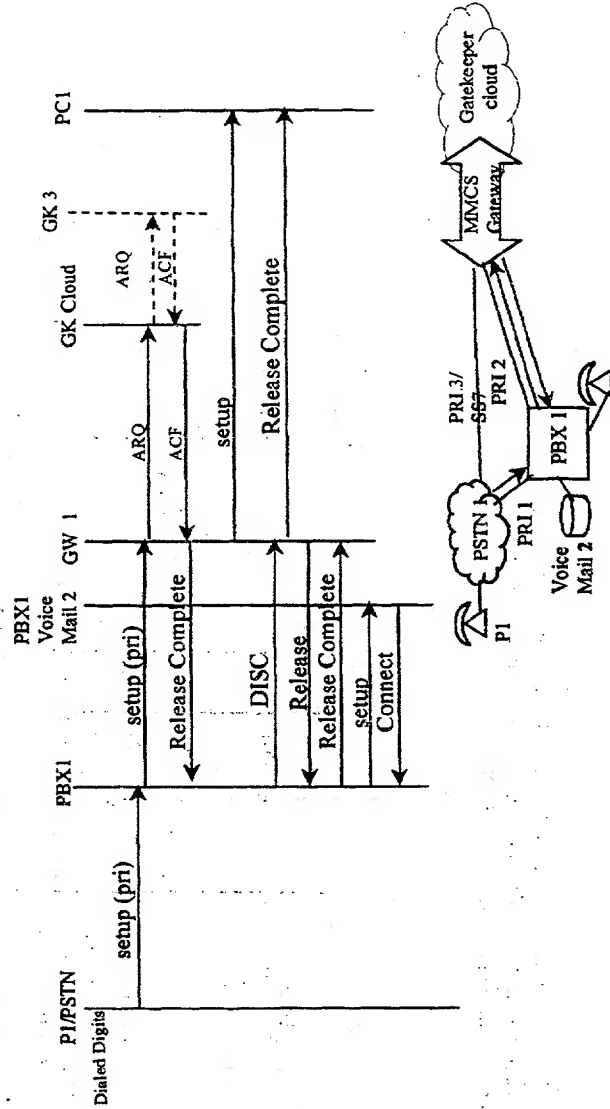


P1 to P8 (voice mail on PBX1 - express mail)

Call specifics.

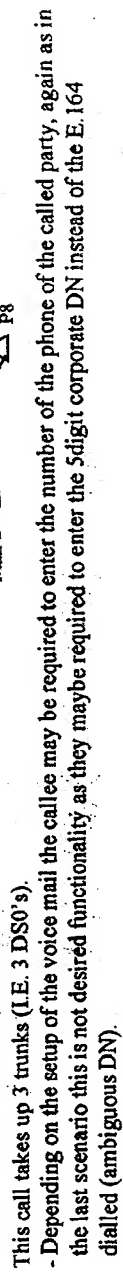
Call from P1 to P8 (phone on PBX1). P8 is call forwarded to PC1 which is not answering Voice Mail 2 on PBX1.

OPTION2: PBX call times out a sends DISCONNECT



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Call specifics. Call from P1 to P8 (phone on PBX1). P8 is call forwarded to PC1 which is not registered. If the PC1 cannot be reached so the gatekeeper is provisioned to send calls to the DN on P8. Essentially this equivalent to P8 and PC1 forwarded to each other and the setup messages could potentially bounce until CP resources are exhausted unless detected (Need to put this in a testcase)

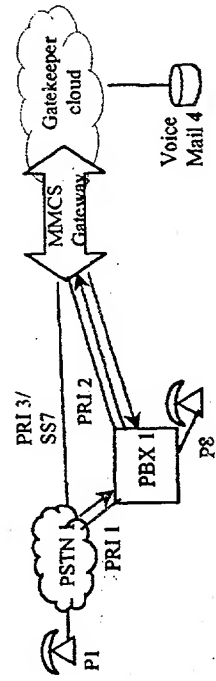
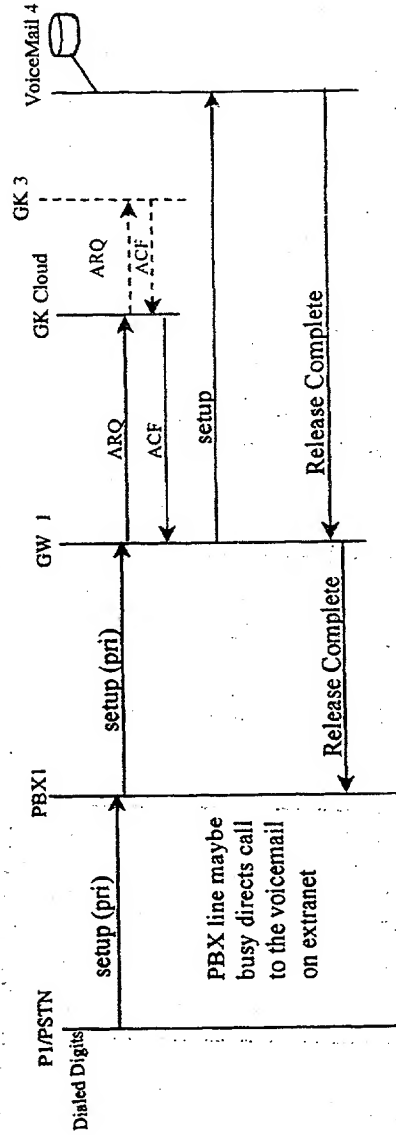


This call takes up 3 trunks (I.E. 3 DS0's).

P1 to P8 (voice mail on Extranet)

Call specifics.

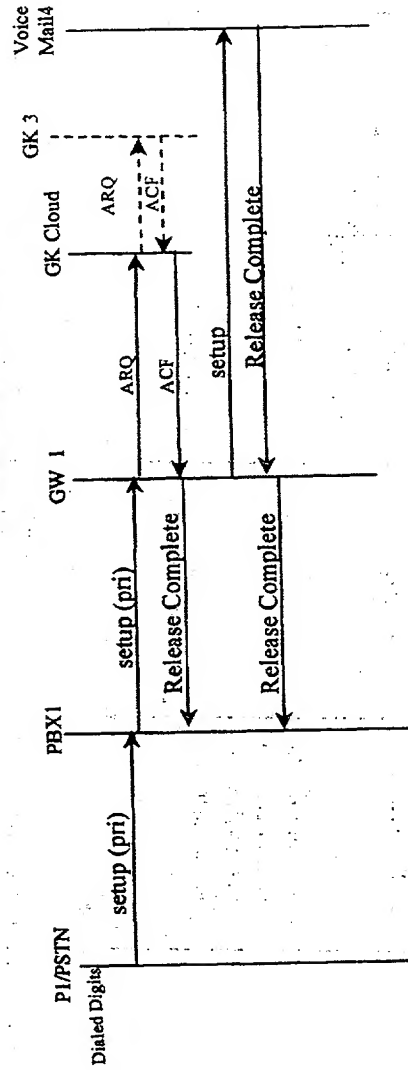
Call from P1 to P8 (phone on PBX1). Voice Mail 4 on extranet.



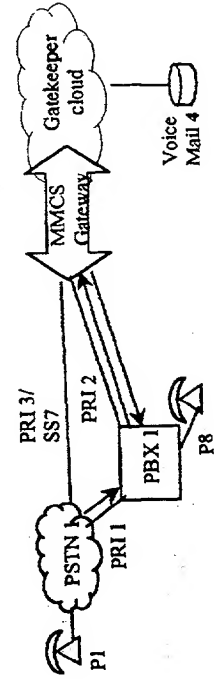
P1 to P8 (voice mail on Extranet)

Call specifics.

Call from P1 to P8 (phone on PBX1). P8 is call forwarded to **PC1** which is **not registered**. Voice Mail 4 on extranet



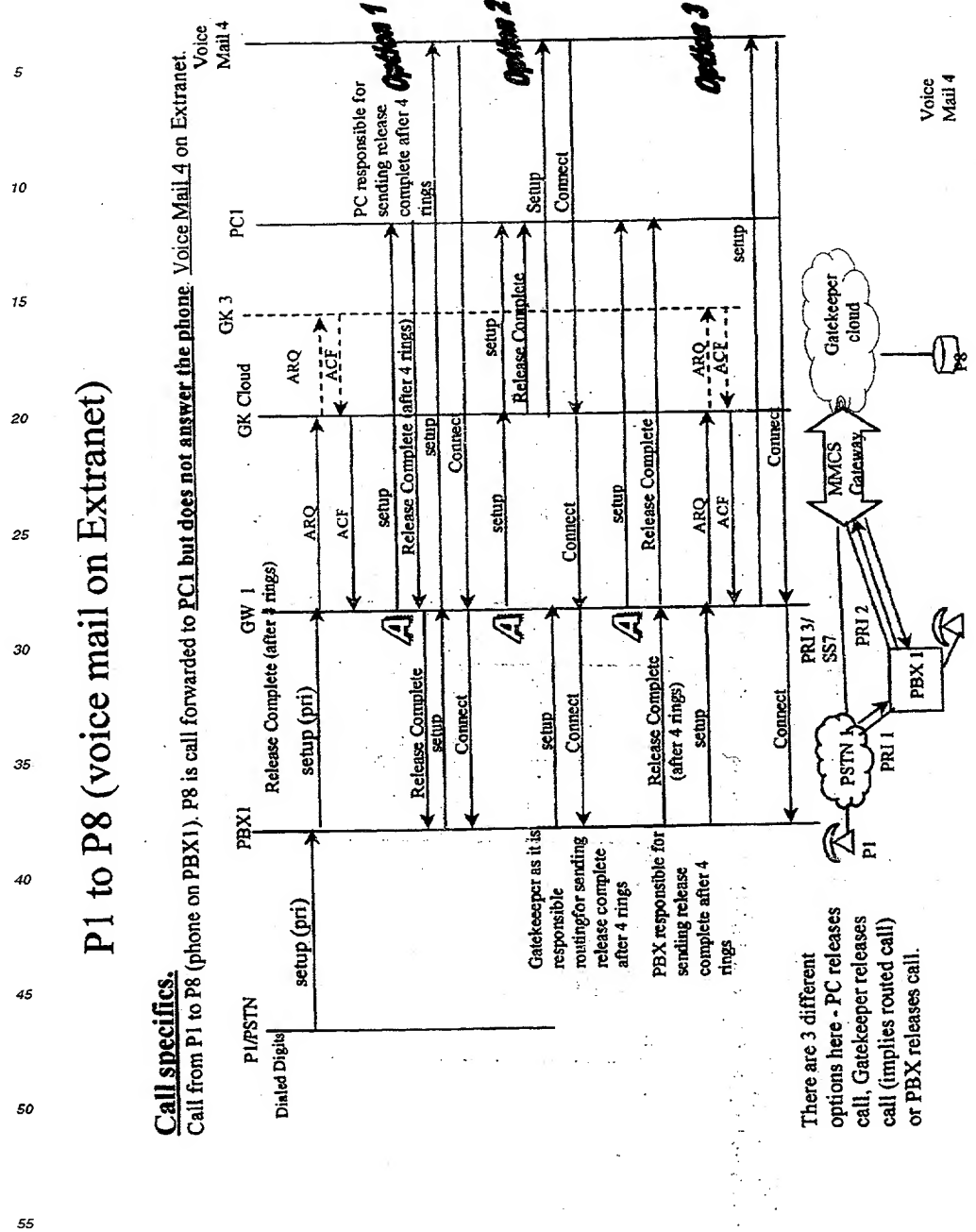
Gatekeeper routes call to VoiceMail 4



P1 to P8 (voice mail on Extranet)

Call specifics.

Call from P1 to P8 (phone on PBX1). P8 is call forwarded to PC1 but does not answer the phone. Voice Mail 4 on Extranet.



There are 3 different options here - PC releases call, Gatekeeper releases call (implies routed call) or PBX releases call.

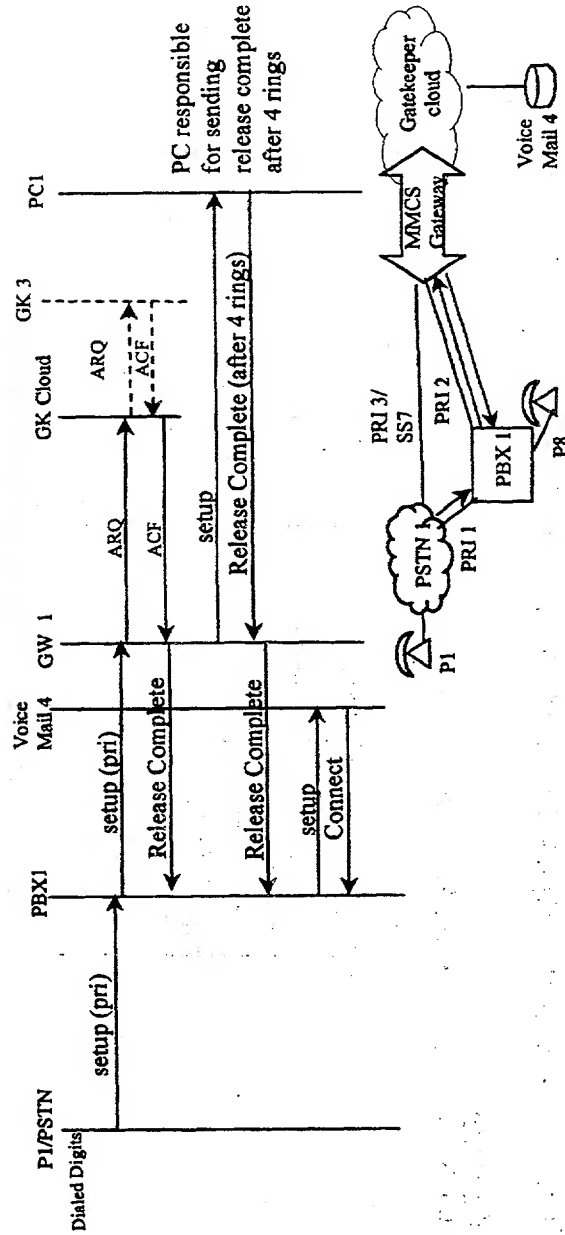
Call specifics. Call from P1 to P8 (phone on PBX1). P8 is call forwarded to **PC1** which is **busy**. Voice Mail 4 on Extranet.



P1 to P8 (voice mail on Extranet)

Call specifics.

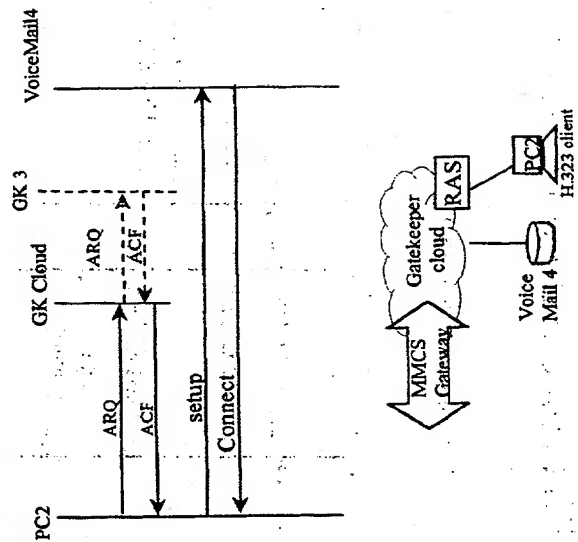
Call from P1 to P8 (phone on PBX1). P8 is call forwarded to PC1 but does not answer the phone. Voice Mail 4 on Extranet.



Calling PC1 via gateway or within extranet (voice mail on Extranet)

Call specifics.

Call to PC1 but **PC1 is not registered**. Voice Mail 4 on Extranet.



Calling PC1 via gateway or within extranet (voice mail on Extranet)

Call specifics.

Call to PC1 but PC1 is connected to voiceMail, Voice Mail 4 on Extranet.

For these scenarios there are 2 options:

- 1) The gatekeeper could route the call and handles all call processing for call setup and release (i.e. checking if PC1 is not answering or busy then routing call to voice mail4. This requires work in Gatekeeper.

- 2) Or use the call forwarding scenarios (CFU/CFB/CF not registered page in slides). The Served (node responsible for call forwarding, normally a gatekeeper) forward calls to voice mail. This also requires work in gatekeeper or PC client depending which node is the served.

IP TELEPHONY GATEWAY APPENDIX 4
Mapping between Q931 parameters
and the
H225/ARQ parameters

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Mapping Q931 parms to H225/ARQ parms

Q.931 on PSTN

H225/ARQ message

requestSeqNum

callType use point-to-point default

callModel, Direct or gatekeeper routed

endpointIdentifier G17, GK or terminal

destinationInfo E, 164 called number

destCallSignalAddress transport address used at the destination for call

signaling

destExtraCallInfo

srcInfo E, 164 calling number

srcCallSignalAddress - transport address used at the source for call signaling

bandwidth - the number of 100 kbps requested for the bi-directional call

callReferenceValue - the CR from Q.931 for this call; only local valid

This is used by a gatekeeper to associate the LRQ with a particular call

nonStandardData - carries information not defined in this recommendation

(for example, proprietary data)

callServices - provides information on support of optional Q-series protocols

to gatekeeper and called terminal

conferenceID - unique conference identifier

activeMC - if TRUE, the calling party has an active MC; otherwise FALSE

answerCall - used to indicate to a gatekeeper that a call is incoming

callMapAlias TRC E, indicates if T containing destinationInfo, destExtraCallInfo

and/or remoteExtension fields, can be copied this information to the same

fields in SETTP message respectively

callIdentifier - a globally unique call identifier set by the originating endpoint

which can be used to associate RLS signaling with the modified Q.931

signaling used in H.225.0

srcAlternatives - prioritized source endpoint alternatives for srcInfo

srcCallSignalAddress, or rasAddress

destAlternatives - a sequence of prioritized destination endpoint alternatives for

destinationInfo or destCallSignalAddress

gatekeeperIdentifier gatekeeperIdentifier received in the alternateGatekeeper

list in RCV

integrityCheckValue encryption requirements

transportQOS indicates QOS reservations done at endpoint, GK or none

willSupplyUIEs set to False if the gatekeeper does not require to see

all UUIEs call control messages

called party

calling party

bearer capability

call reference may be not the same as on the Gateway

3WC call ?

Conference Call using Multicasting

Not a Call reference

alternative calling party not part of Q.931

Mapping Q931 parms to H225/ARQ parms

Q.931 on PSTN

H225/ACF message

requestSeqNum - This shall be the same value that was passed in the IRR.
bandWidth - the allowed maximum bandwidth for the call; may be less than that requested. → bearer capability

callModel - tells terminal whether call signaling sent on dest 'allSignalAddress goes to a gatekeeper routed call or to a terminal/direct call.

destCallSignalAddress - the transport address to which to send Q.931 call signaling. But may be an endpoint or gatekeeper address depending on the call model in use.

irrFrequency - the frequency, in seconds, that the endpoint shall send IRRs to the gatekeeper while on a call, including while on hold. If not present, the endpoint does not send IRRs while active on a call, and it is expected that the gatekeeper will poll the endpoint.

nonStandardData - carries information not defined in this recommendation (for example, proprietary data)

destinationInfo - the address of the initial channel, used when calling through a gateway.

destExtraCallInfo - needed to make possible additional channel calls, i.e. for a 2*64 Kbps call on the H:AN side. Shall only contain E.164 addresses and shall not contain the number of the initial channel.

destinationType - This specifies the type of the destination endpoint i.e. gatekeeper, gateway, net, or terminal

remoteExtensionAddress - contains the alias address of a called endpoint in cases where this information is needed to traverse multiple gateways

alternateEndpoints - a sequence of prioritized endpoint alternatives

destCallSignalAddress or destinationInfo

tokens - This is some data which may be required to allow the operation. The data shall be inserted into the message if available.

cryptoTokens - encrypted tokens

integrityCheckValue - cryptographically based integrity check value

TransportQOS - Gatekeeper may indicate to the endpoint responsible for resource reservation.

willRespondToIRR - true if the Gatekeeper will send an IRR or IN:K message in response to an unsolicited IRR message when the IRR's needsResponse field set to true.

uriesRequested indicates the set of H.225.0 call signaling messages of which the endpoint shall notify the gatekeeper.

SETUP UUIE message

H.225/Q.931
Setup header

Q.931 from
PSIN

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protocolIdentifier H.225 version
h245Address transport address on which the calling endpoint or gatekeeper handles establish of H.245 signaling. Sender is capable of handling H.245 procedures before receiving a CUXNSET on the Call Signaling channel.

sourceAddress alias addresses for source I.F.F. 164 number Q.931 Calling Party Number IE, sourceInfo contains an EndpointType GIV GIK etc

destinationAddress E.164 address, same as Q.931 Called Party Number IE, if available, include in the Setup message by version 2 terminals.

destCallSignalAddress - inform the gatekeeper of the destination terminal's call signaling transport address, redundant in the direct terminal-to-terminal case. If available must be filled in.

destExtncCallInfo additional channel calls, i.e. for a 2*64 Kbps on the H.245 side. Contain E.164 addresses

destExtncRV - CRI's for the additional SNA calls specified by destExtncCallInfo. Their use is for further study.

activeMC - Calling endpoint is under the influence of an active MC

conferenceID - unique conference identifier

conferenceGoal

callIndependentSupplementaryService - transport of supplementary services. IPDI's in a non-call related manner

callServices - provides information on support of optional Q-series protocols to gatekeeper and called terminal.

callType - default value is pointToPoint for all calls

sourceCallSignalAddress transport address for the source. Used in the ARQ message by the receiver of the Setup

remoteExtensionAddress alias address of a called endpoint. When needed to traverse multiple gateways.

callIdentifier - a globally unique call identifier set by the originating endpoint which can be used to associate RIS signaling with the modified Q.931 signaling used in H.225.0

h245SecurityCapability - a set of capabilities the sender can use to secure the H.245 channel tokens. This is some data which may be required to allow the operation. The data shall be inserted into the message if available.

cryptoTokens - encrypted tokens

fastStart - Used only in the fast connect procedure. fastStart supports the signaling needed to open a logical channel. I.E. OpenLogicalChannel structure defined in H.245. Sender indicates preferred mode Rx Tx, transport addresses where it expects to receive media streams.

mediaWaitForConnect - If TRUE, indicates that the recipient of the Setup message shall not transmit media until sending the Connect message.

canOverlapSend - If TRUE, sender of Setup shall support overlap sending (set to false)

callingparty IE

calledparty

callingparty IE

calledparty

Note: In the ARQ message srcInfo, destinationInfo are equivalent to the SETUP UUIE sourceAddress, destinationAddress respectively.

Detailed Call Flow

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Registration Accepted

Call specifics: PCI registers with gatekeeper (registration accepted)

PCI

RAS

GK

Road Warrior connects to RAS with his unique identifier (This will unlikely be an E.164 address as in GSM). Sends a call reference for this call.

RRQ (parms: RequestSeqNum, endpoint, terminalalias, terminaltype, gatekeeperIdentifier, callSignalAddress)

The terminal alias contains the E.164 address for unique identifier for the PCI. Check authorization to ensure E.164 number is valid.

The E.164 and other alias address (i.e. unique identifier) are contained in the terminalAlias field. terminalType is GK, GW or terminal. The endpointVendor could be PCI or i.e. netmeeting.

Using the uniqueID, we could possibly validate if the user is authentic.

RCF (parms: RequestSeqNum, callSignalAddress, terminalAlias, gatekeeperIdentifier, alternateGatekeeper, preGrantedARQ)

The Road Warrior receives authentication.

Other parameters indicate when to use ARQ or not

Note 1: An E.164 address is location specific. How do we support a single DN across the PSTN and IP network. This can only be done using the GSM idiom.

Note 2: I have underlined parameters that are of interest to us for supplementary services.

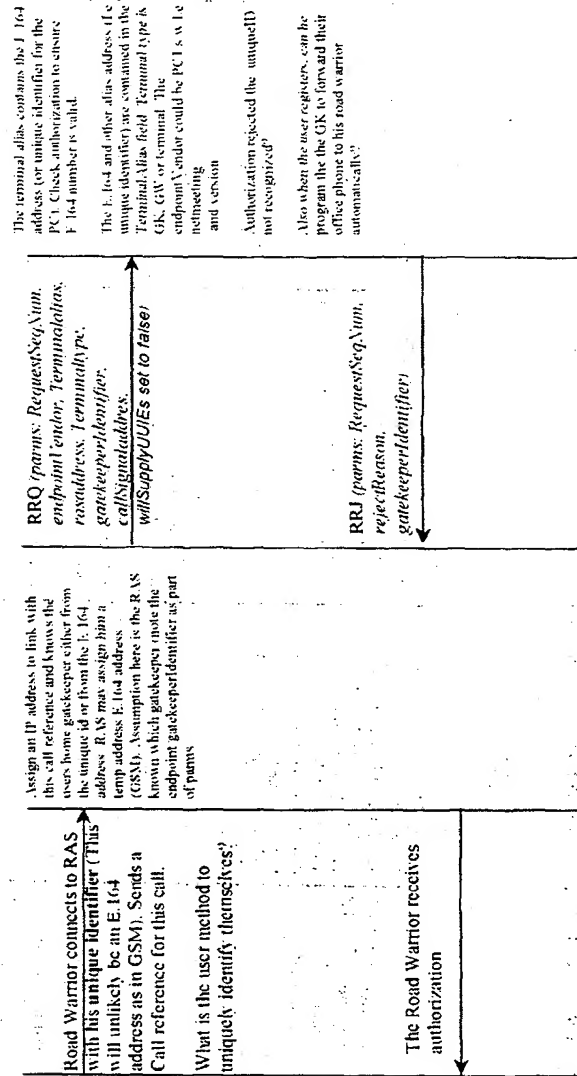
Registration Rejected

Call specifics: PCI registers with gatekeeper (registration rejected by gatekeeper)

GK

RAS

PCI

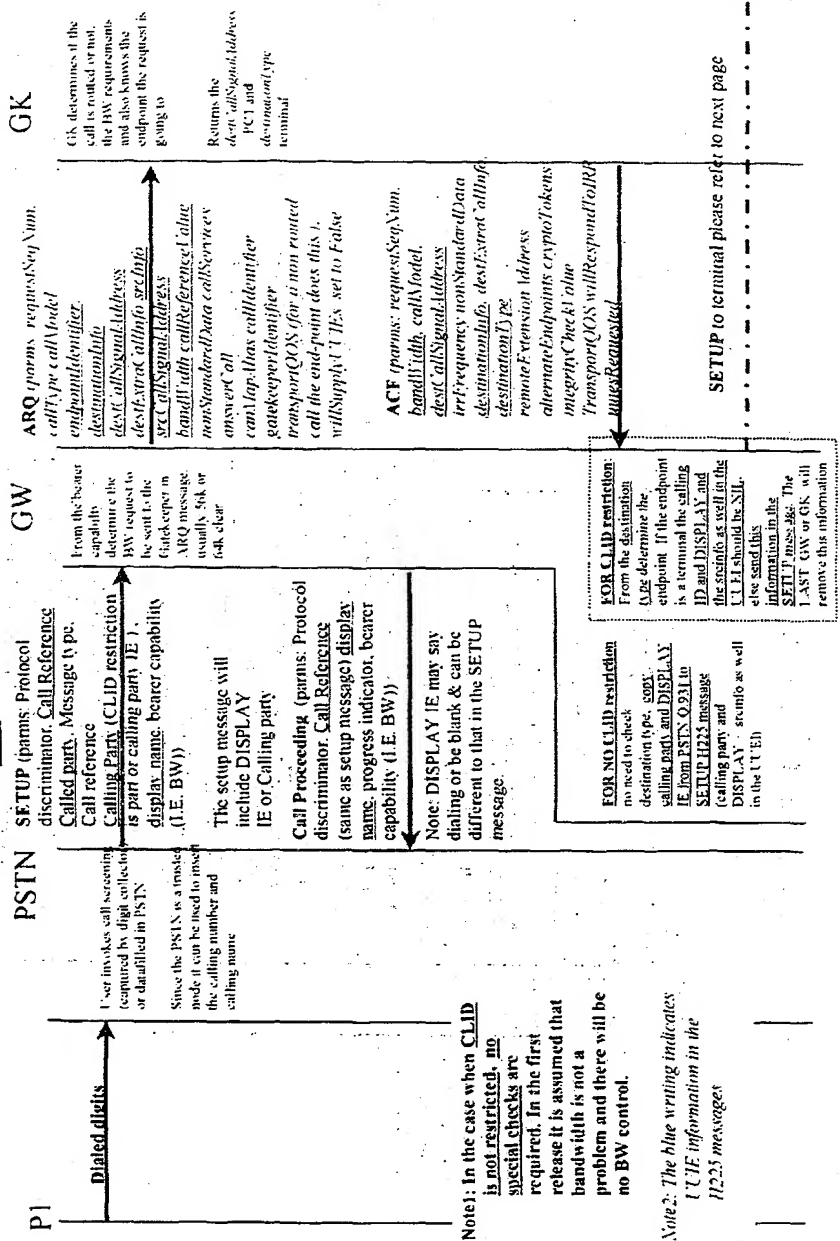


Note: An E.164 address is location specific.

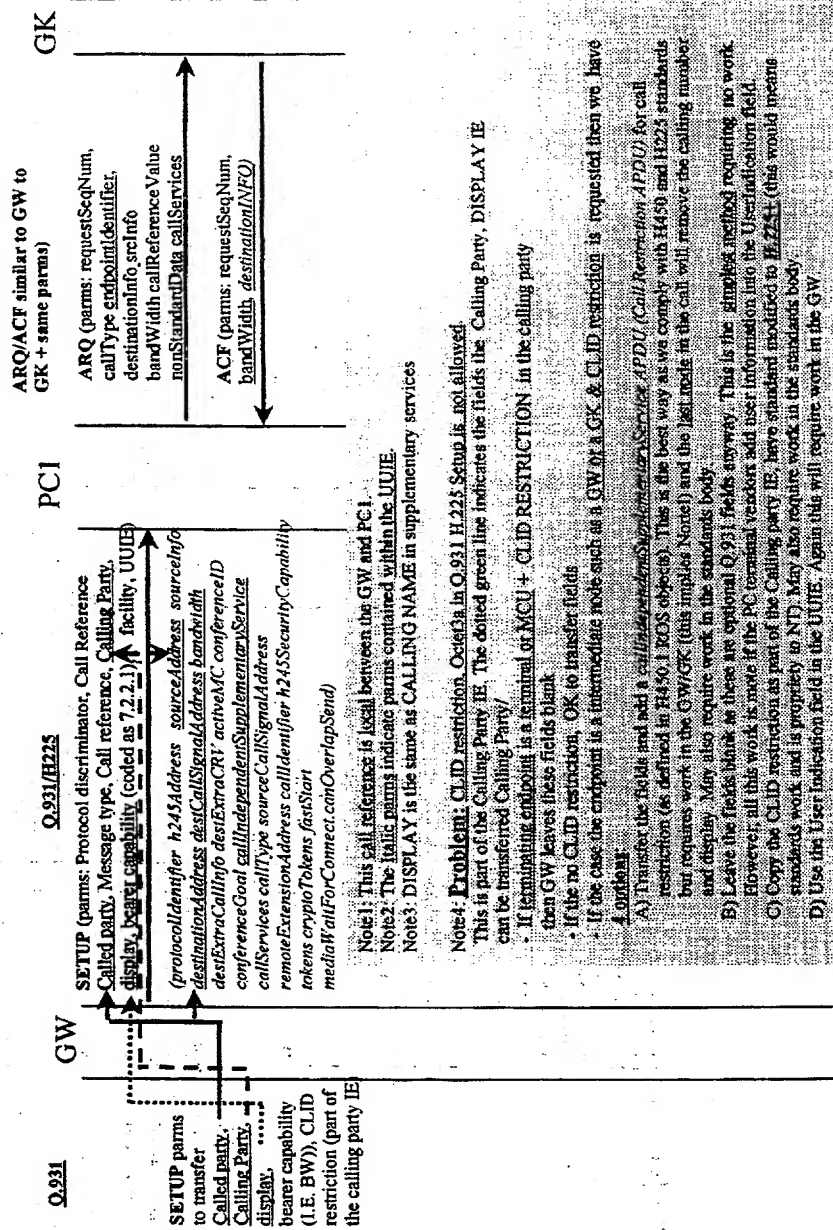
How do we support a single-DN across the PSTN and IP network. This can only be done using the GSM idiom of assigning a temporary E.164 (done by RAS). User device has 3 ids, its own unique, one assigned by RAS and an IP assigned by RAS. All must be sent to the GK.

P1 calls PC1 - Page 1

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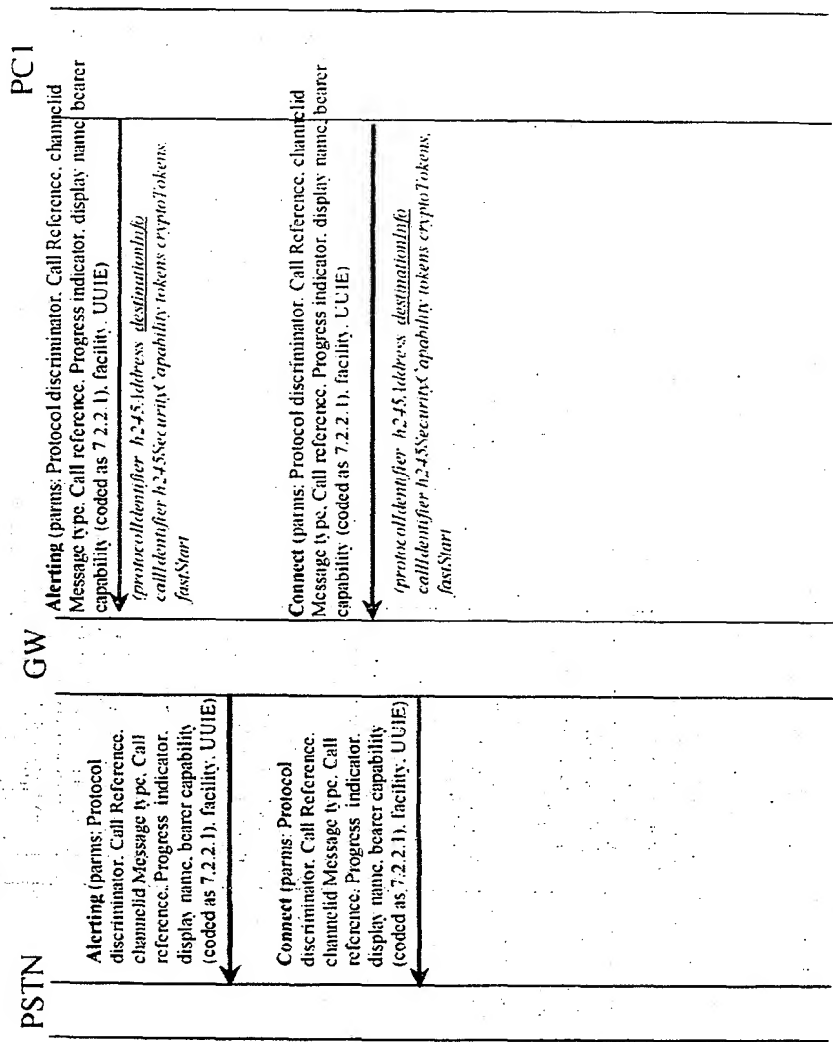


P1 calls PC1 - Page 2

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P1 calls PCI - Page 3





P1 PSTN1 calls P5 PSTN 2 - Page 3

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P5

PSTN2

Q931

GW2

Q931/H225

GW

Facility: call rerouting

Facility: call rerouting return result

Release Complete

SETUP

Ring ing

Note: The GW2 may forward to another card?
Is there a concept of leader follower cards on Trunk side?

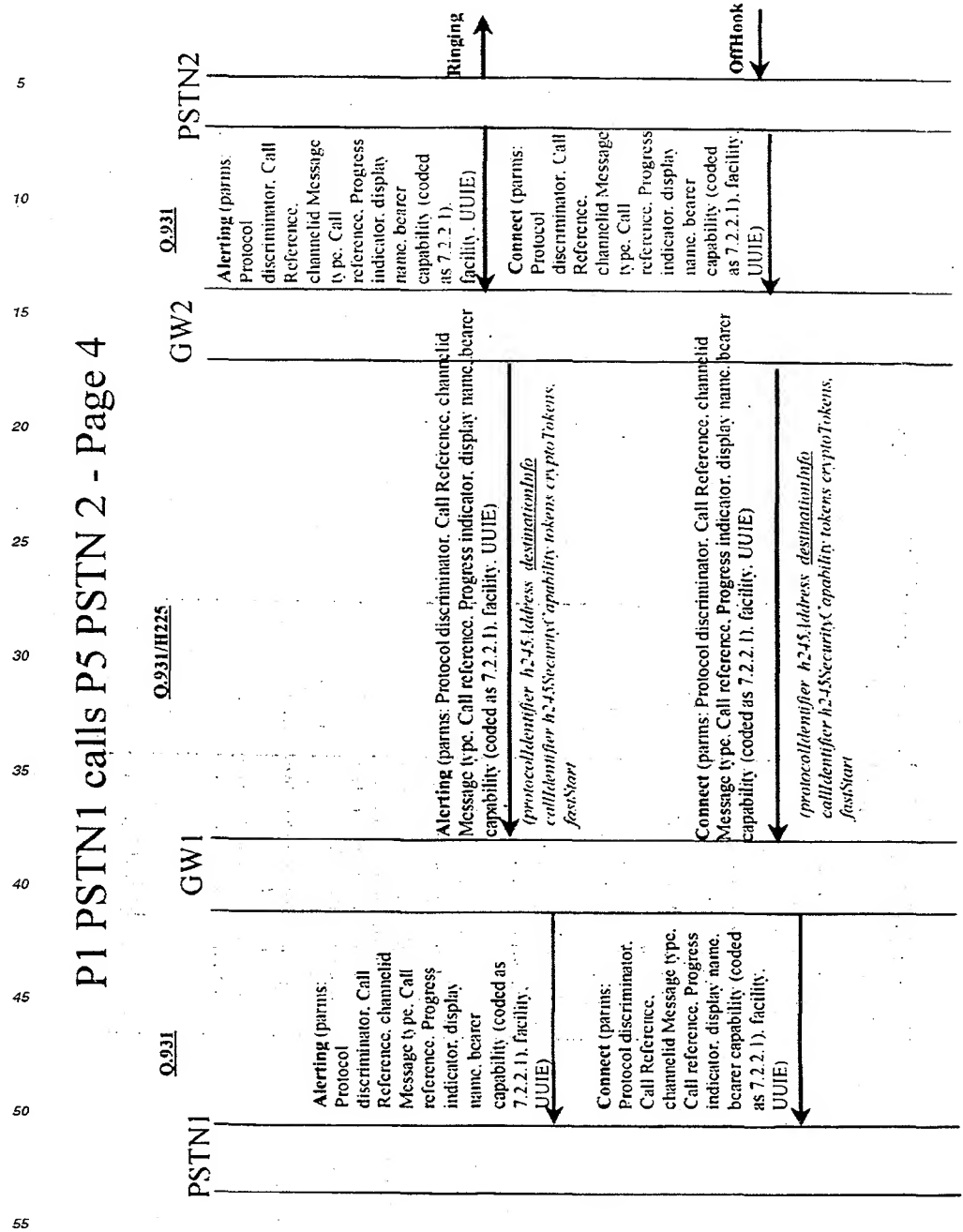
If the PSTN receives the CLID restriction then it must blank calling name and calling number

Depending on the 4 options indicated on the previous page, Gateway 2 can do the following:

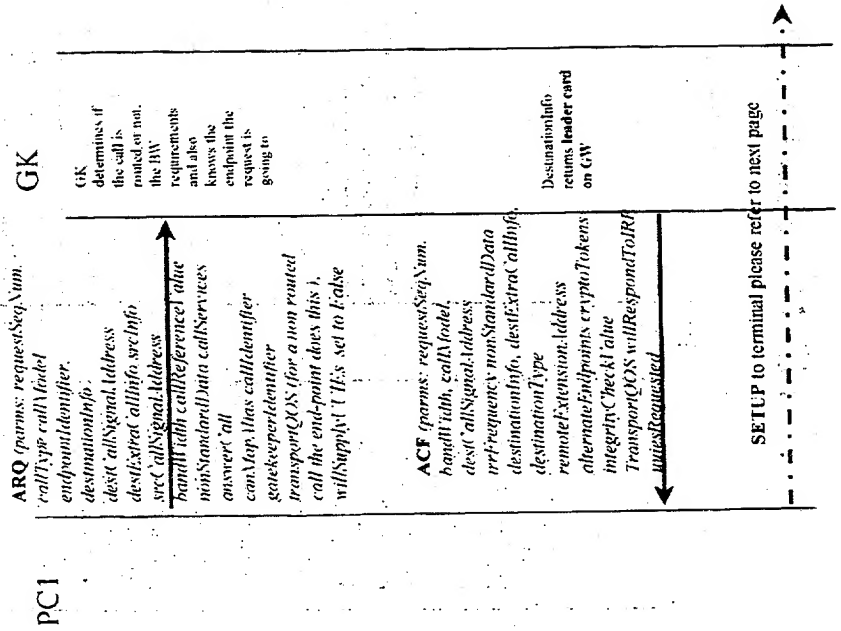
- A) Transfer the fields, and add a call independent supplementary service (CIS) if call restriction (H225).
- B) GW 2 has the corresponding APD implemented and reads the CLID restriction field to the Q931 setup message (calling party IE). This is the most difficult of all the options but the most standards based.
- C) Leave the fields blank as these are optional Q931 fields anyway. This is the simplest method requiring no work.
- D) No work on GW2. CLID restriction is required to be added.
- E) Copy the CLID restriction as part of the Calling party IE. Have standard modified to H225.
- F) Minimum work to copy fields across from H225 to Q931.
- G) Use the User Indication field in the CT IE. Again this will require work in the GW.
- H) Work required at the GW. Still no standards based and could be a limitation on the number of characters that can be placed in the User Indication field.

I would Recommend Option B for simplicity

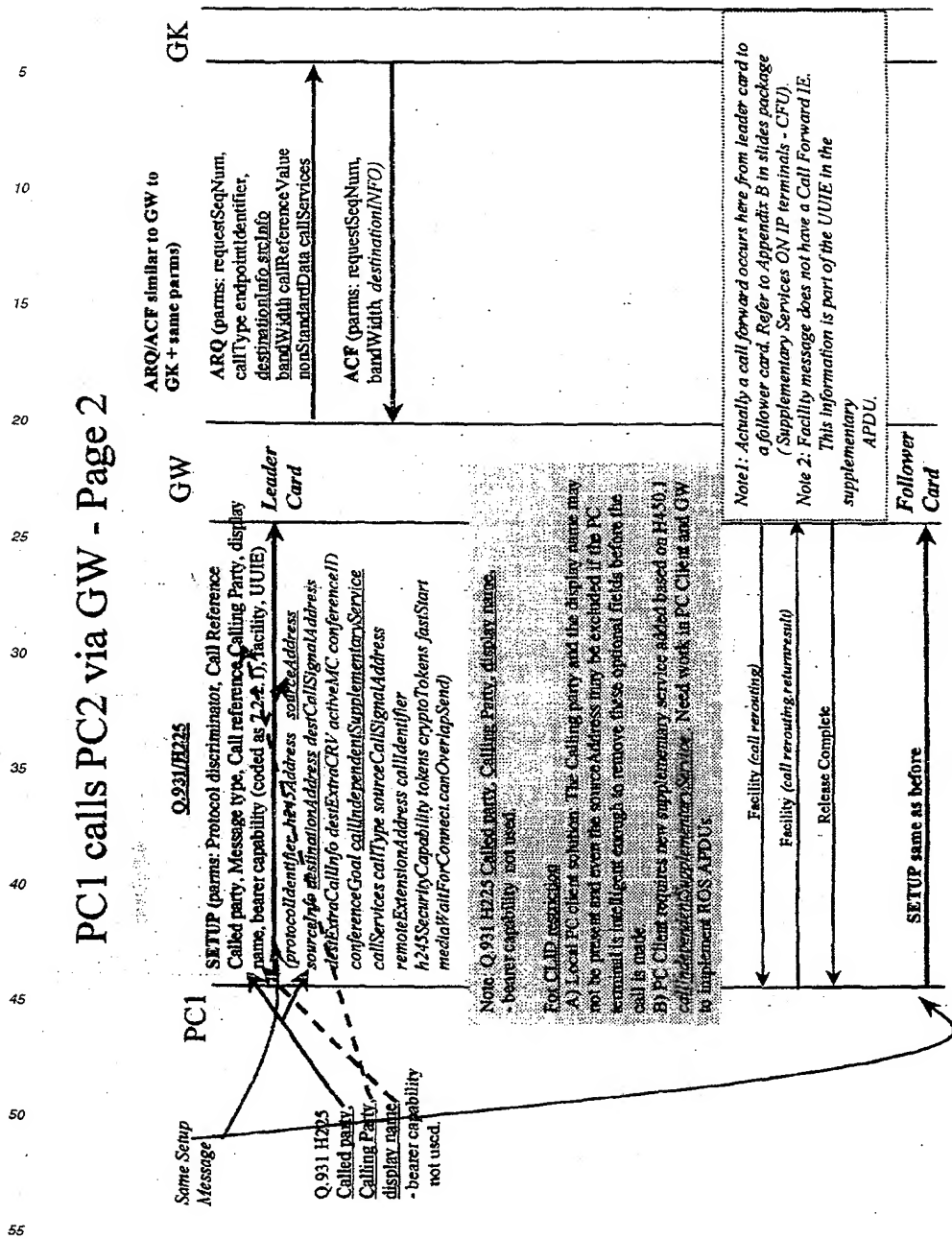
P1 PSTN1 calls P5 PSTN 2 - Page 4



PC1 calls PC2 (IP terminal to IP terminal) via GW- Page 1 Call specifics: The GW is used to make use of Billing records software available in the MMCS.



PC1 calls PC2 via GW - Page 2



PC1 calls PC2 via GW - Page 3

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PC2

Q.931/H.225

SETUP (parms: Protocol discriminator, Call Reference
Called party, Message type, Call reference Calling Party, display
name, bearer capability (coded as 7.2.2.1), facility, UIIE)

(protocolIdentifier h245, address source, address
sourceInfo destination, address dest, callSignal, address
destExt, callInfo destExt, 'R' active, 'C' confer, 'e' l)
conference, callType, source, callSignal, address
remoteExtension, address callIdentifier
h245SecurityCapability, tokens crypto, tokens lastStart
media, callInfo, 'n'ice, 't'ran, 'o'p, 's'ub)

Alerting (parms: Protocol discriminator, Call Reference, channelid
Message type, Call reference, Progress indicator, display name, bearer
capability (coded as 7.2.2.1), facility, UIIE)

(protocolIdentifier h245, address destinationInfo,
callIdentifier h245SecurityCapability, tokens crypto, tokens
fastStart)

Connect (parms: Protocol discriminator, Call Reference, channelid
Message type, Call reference, Progress indicator, display name, bearer
capability (coded as 7.2.2.1), facility, UIIE)

(protocolIdentifier h245, address destinationInfo,
callIdentifier h245SecurityCapability, tokens crypto, tokens
fastStart)

GW

Q.931/H.225

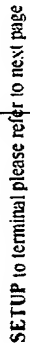
Follower
Card

Alerting (parms: Protocol
discriminator, Call Reference,
channelid Message type, Call
reference, Progress indicator,
display name, bearer capability
(coded as 7.2.2.1), facility, UIIE)

Connect (parms: Protocol
discriminator, Call Reference,
channelid Message type, Call
reference, Progress indicator,
display name, bearer capability
(coded as 7.2.2.1), facility, UIIE)

PC1

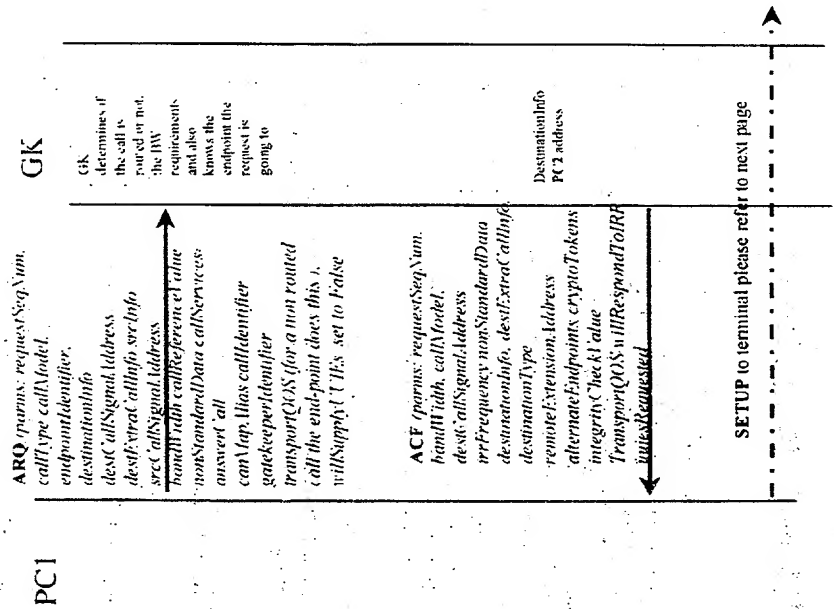
Call specifics: This is a POT'S phone that is connected to an IP adjunct on the IP extranet



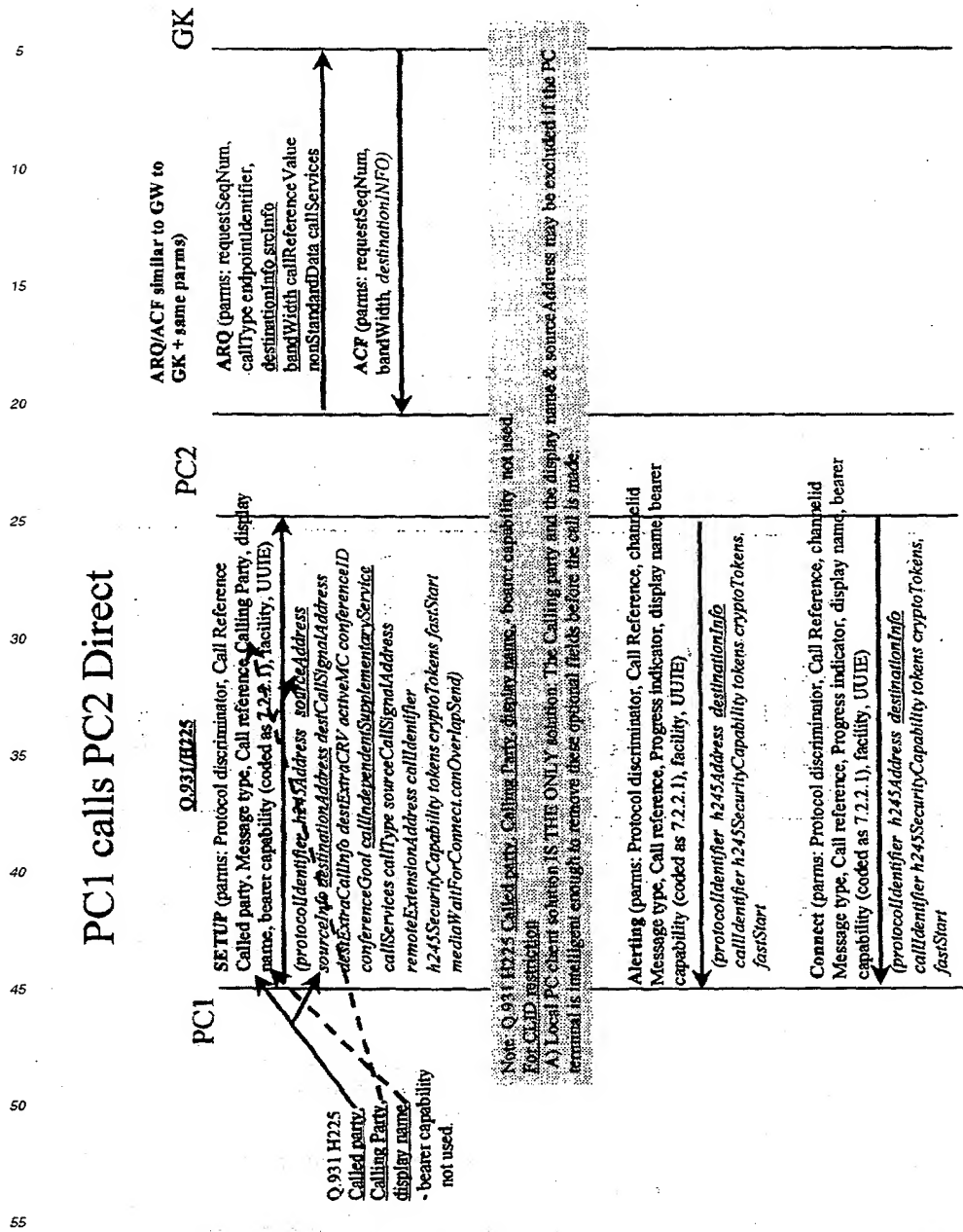


PC1 calls PC2 DIRECT - Page 1

Note: How do we handle Billing if not through the Gateway?



PC1 calls PC2 Direct



Claims

- 5 1. A gateway for use between an IP network and another network, the gateway being adapted to handle calls between IP terminal devices connected to the IP network as well as calls between an IP terminal device and a terminal device connected to the other network, the gateway being further adapted to provide at least one supplementary service for calls to or from an IP terminal device.
- 10 2. The gateway according to claim 1, wherein the supplementary service is chosen from at least one of:
 - a terminating restriction;
 - call forwarding;
 - calling line identification;
 - CLID restriction;
 - 15 - calling name display;
 - call transfer.
- 20 3. The gateway according to any previous claim, wherein the gateway is adapted to provide the supplementary service on a call between two IP terminal devices and/or to provide the supplementary service on a call between an IP terminal device and a terminal device connected to the other network.
- 25 4. The gateway according to any previous claim, wherein the gateway comprises a shared pool of ports on the line side which are usable for a connection to an IP terminal device.
5. The gateway according to any previous claim, wherein the gateway is adapted to dynamically associate an IP terminal device client's subscriber data with a call.
- 30 6. The gateway according to any previous claim, wherein the gateway is adapted to perform address resolution for calls to IP terminal devices.
7. The gateway according to any previous claim, wherein the gateway is integrated with a switch.
- 35 8. An IP network for connection to another network, the IP network being adapted for handling calls between IP terminal devices connected to the IP network as well as calls between an IP terminal device and a terminal device connected to the other network, the network being further adapted to provide at least one supplementary service for calls to or from an IP terminal device.
- 40 9. The IP network according to claim 8, wherein the supplementary service is chosen from at least one of:
 - a terminating restriction;
 - call forwarding;
 - calling line identification;
 - CLID restriction;
 - 45 - calling name display;
 - call transfer.
- 50 10. The IP network according to claim 8 or 9, wherein the network is adapted to provide the supplementary service on a call between two IP terminal devices and/or is adapted to provide the supplementary service on a call between an IP terminal device and a terminal device connected to the other network.
- 55 11. The IP network according to any of claims 8 to 10, wherein the network is adapted to dynamically associate an IP terminal device client's subscriber data with a call.
12. The IP network according to any of claims 8 to 11, wherein a voice call between two IP terminal devices without double encoding/decoding of the voice data.
13. The IP network according to any of claims 8 to 12, further comprising a gateway, the gateway being adapted to

provide the supplementary service.

- 5 14. The IP network according to any of claims 8 to 13, wherein the gateway comprises a shared pool of ports on the line side which are usable for a connection to an IP terminal device.
15. The IP network according to any of claims 8 to 14, wherein the network is adapted to route call control signals for a call between two IP terminal devices through the gateway or the IP network is adapted to route call control signals for a call between two IP terminal devices through the IP network and call signaling through the gateway.
- 10 16. A method of operating a gateway between an IP network and another network, the gateway being adapted to handle calls between IP terminal devices connected to the IP network as well as calls between an IP terminal device and a terminal device connected to the other network, the method including the step of providing at least one supplementary service for calls to or from an IP terminal device.
- 15 17. The method according to claim 16, wherein the supplementary service is chosen from at least one of:
 originating restrictions;
 - a terminating restriction;
 - call forwarding;
 - 20 - calling line identification;
 - CLID restriction;
 - calling name display;
 - call transfer.
- 25 18. The method according to claim 16 or 17, wherein the supplementary service is provided on a call between two IP terminal devices and/or is provided on a call between an IP terminal device and a terminal device connected to the other network.
- 30 19. The method according to any of the claims 16 to 18, further comprising the step of dynamically associating an IP terminal device client's subscriber data with a call.
- 35 20. A method of operating an IP network connected to another network, the IP network handling calls between IP terminal devices connected to the IP network as well as calls between an IP terminal device and a terminal device connected to the other network, the method comprising the step of providing at least one supplementary service for calls to or from an IP terminal device.
- 40 21. The method according to claim 20, wherein the supplementary service is chosen from at least one of:
 originating restrictions;
 - a terminating restriction;
 - call forwarding;
 - calling line identification;
 - CLID restriction;
 - calling name display;
 - 45 - call transfer.
22. The method according to claim 20 or 21, further comprising the step of dynamically associating an IP terminal device client's subscriber data with a call.
- 50 23. The method according to any of claims 20 to 22, further comprising the step of routing a voice call between two IP terminal devices without double encoding/decoding of the voice data.
- 55 24. A gateway between an IP network and another network, the gateway handling calls between IP terminal devices connected to the IP network as well as calls between an IP terminal device and a terminal device connected to the other network, the gateway comprising a shared pool of ports on the line side which are usable for a connection to an IP terminal device.
25. The gateway according to claim 24, wherein the gateway is adapted to dynamically associate an IP terminal device

client's subscriber data with a call.

26. A method of operating IP network having a gateway between the IP network and another network, the gateway handling calls between IP terminal devices connected to the IP network as well as calls between an IP terminal device and a terminal device connected to the other network, the method including the steps of:
5 routing call signaling for a call between two IP terminals through the gateway and routing voice traffic between two IP terminals without passing via the gateway.
27. An IP network having a gateway between an IP network and another network, the gateway handling calls between
10 IP terminal devices connected to the IP network as well as calls between an IP terminal device and a terminal device connected to the other network, the method including the steps of:
routing call signaling for a call between two IP terminals through the gateway and routing voice traffic between two
IP terminals without passing via the gateway.

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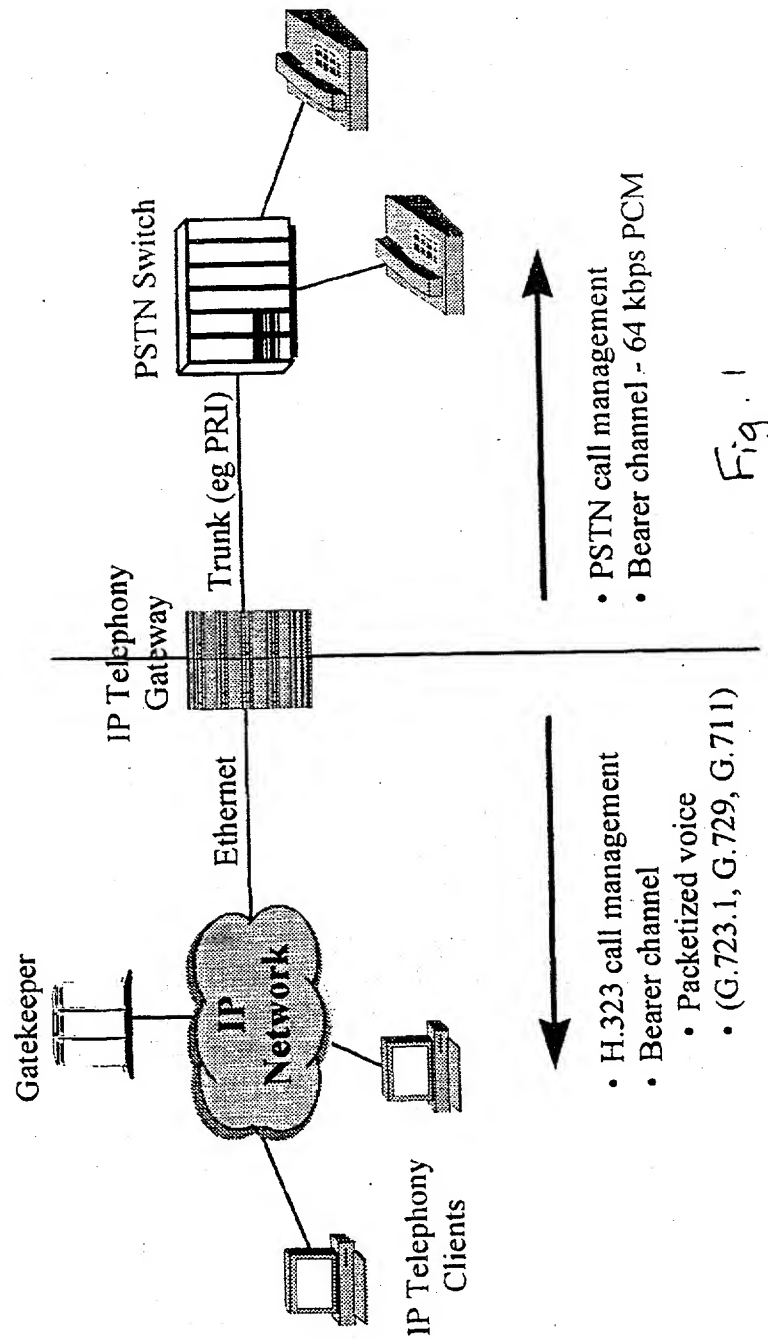


Fig. 1

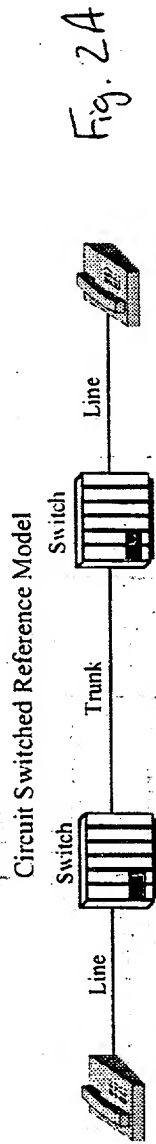


Fig. 2A

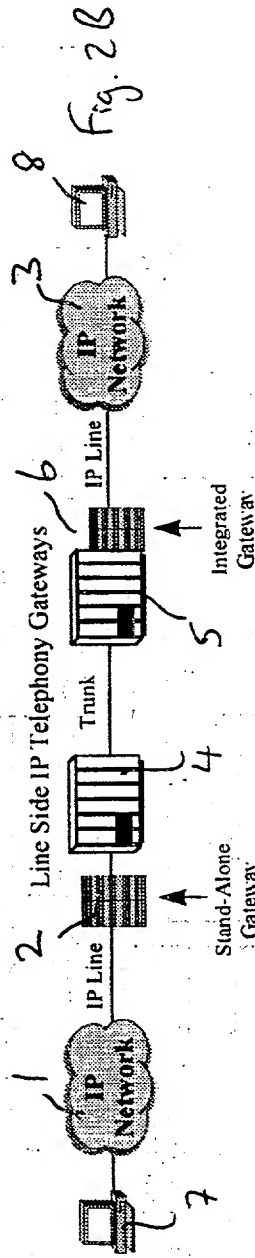


Fig. 2B

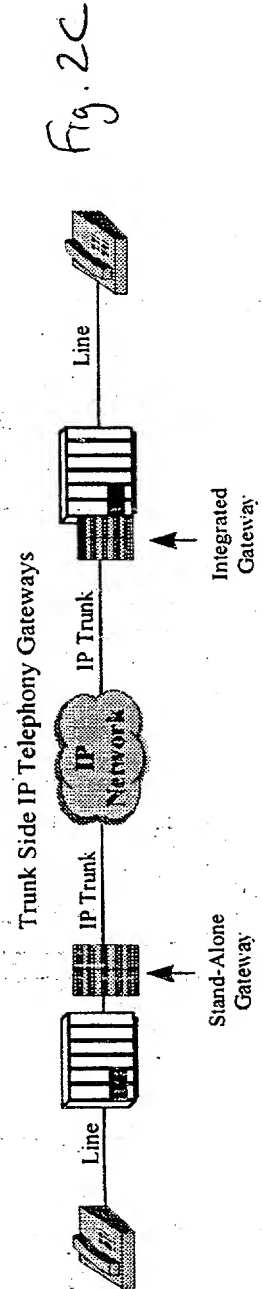


Fig. 2C

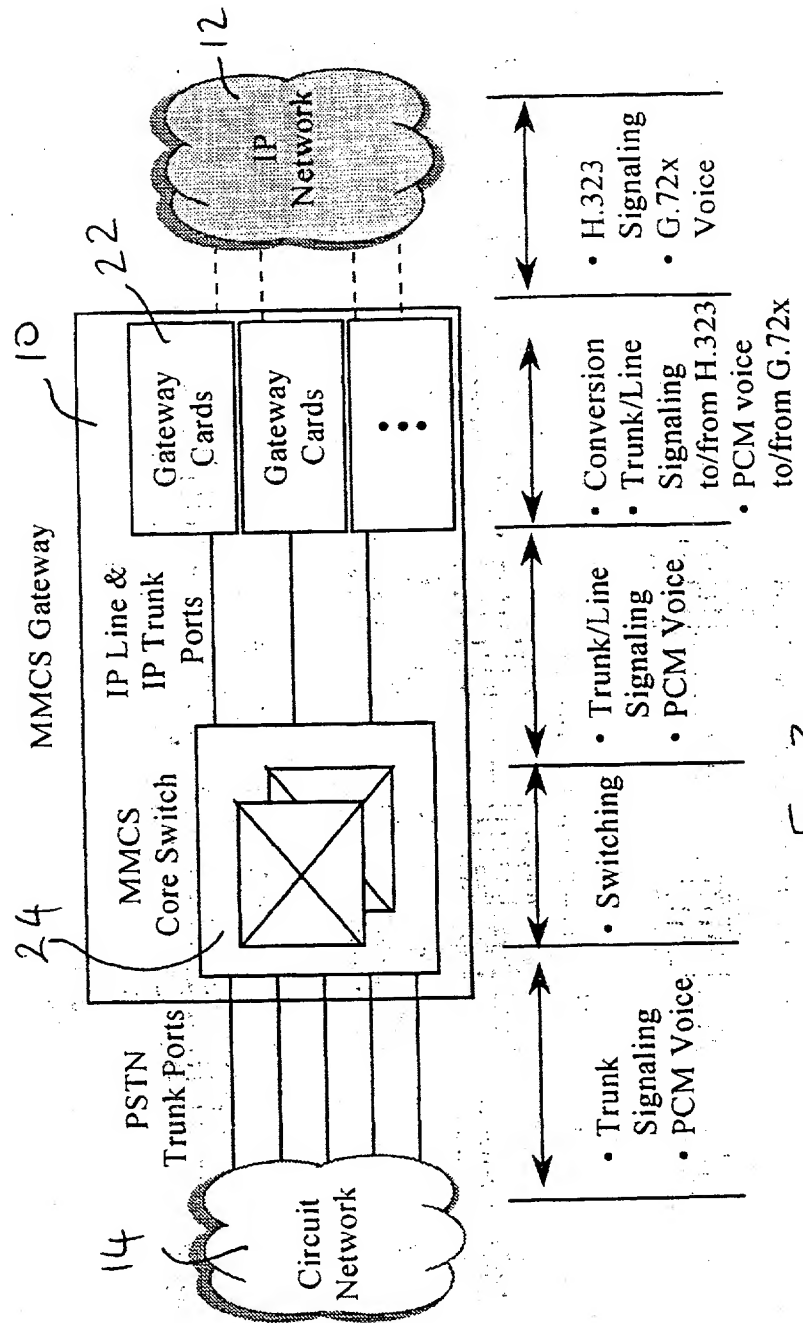


Fig. 3

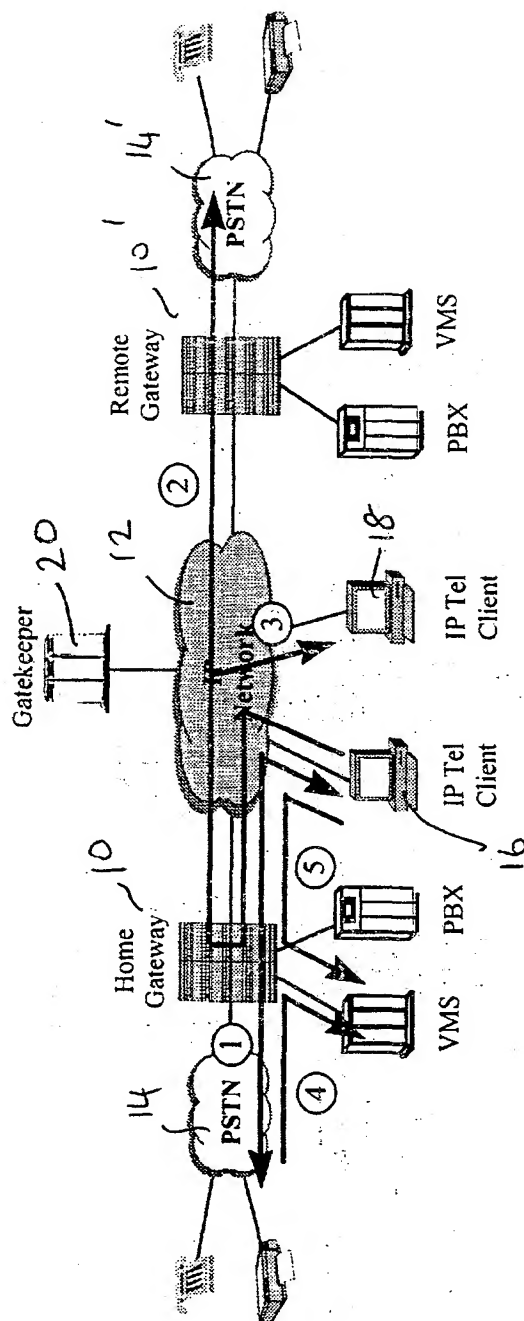


Fig. 4A

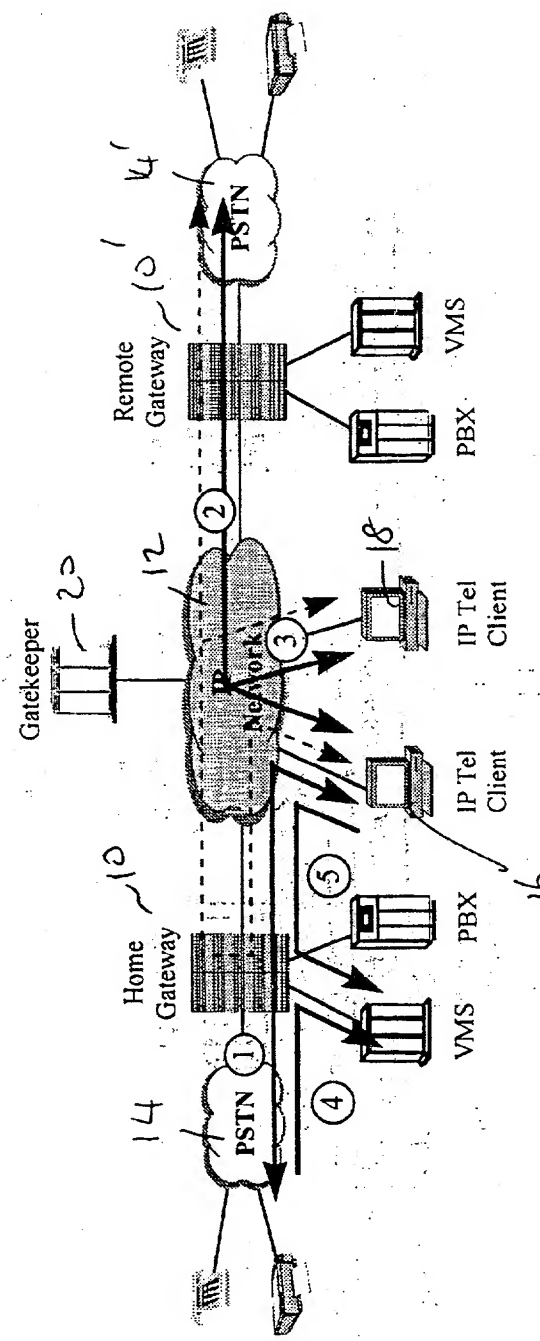


Fig. 4B

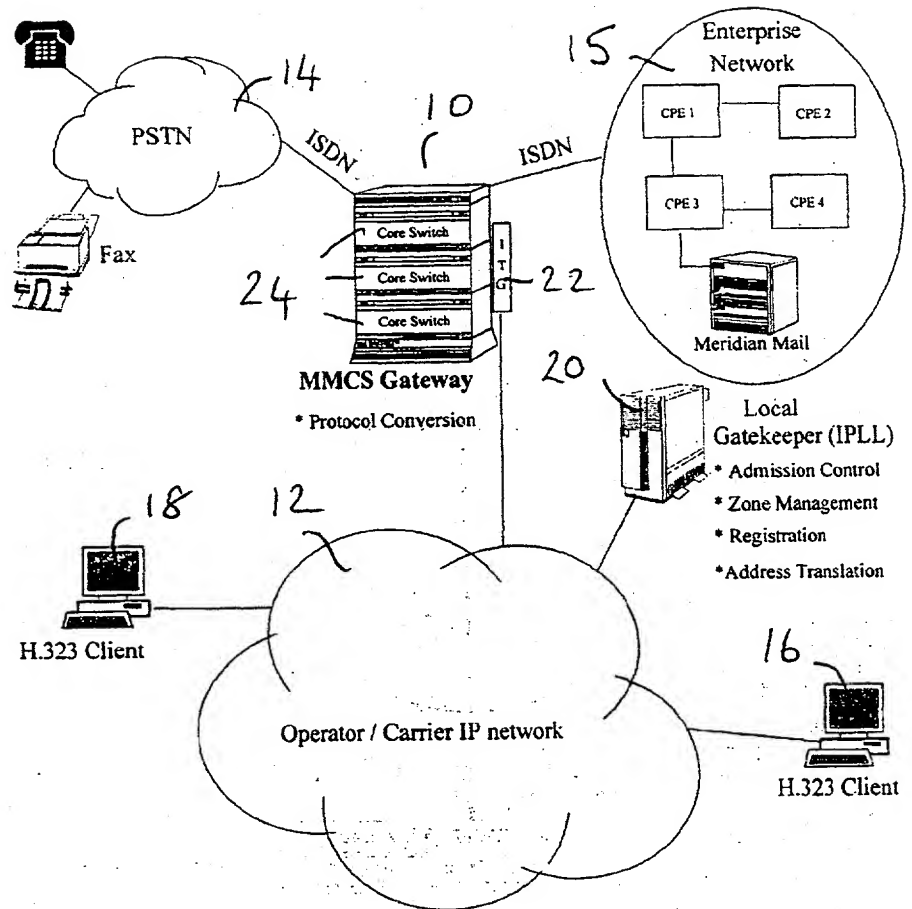


Fig. 5

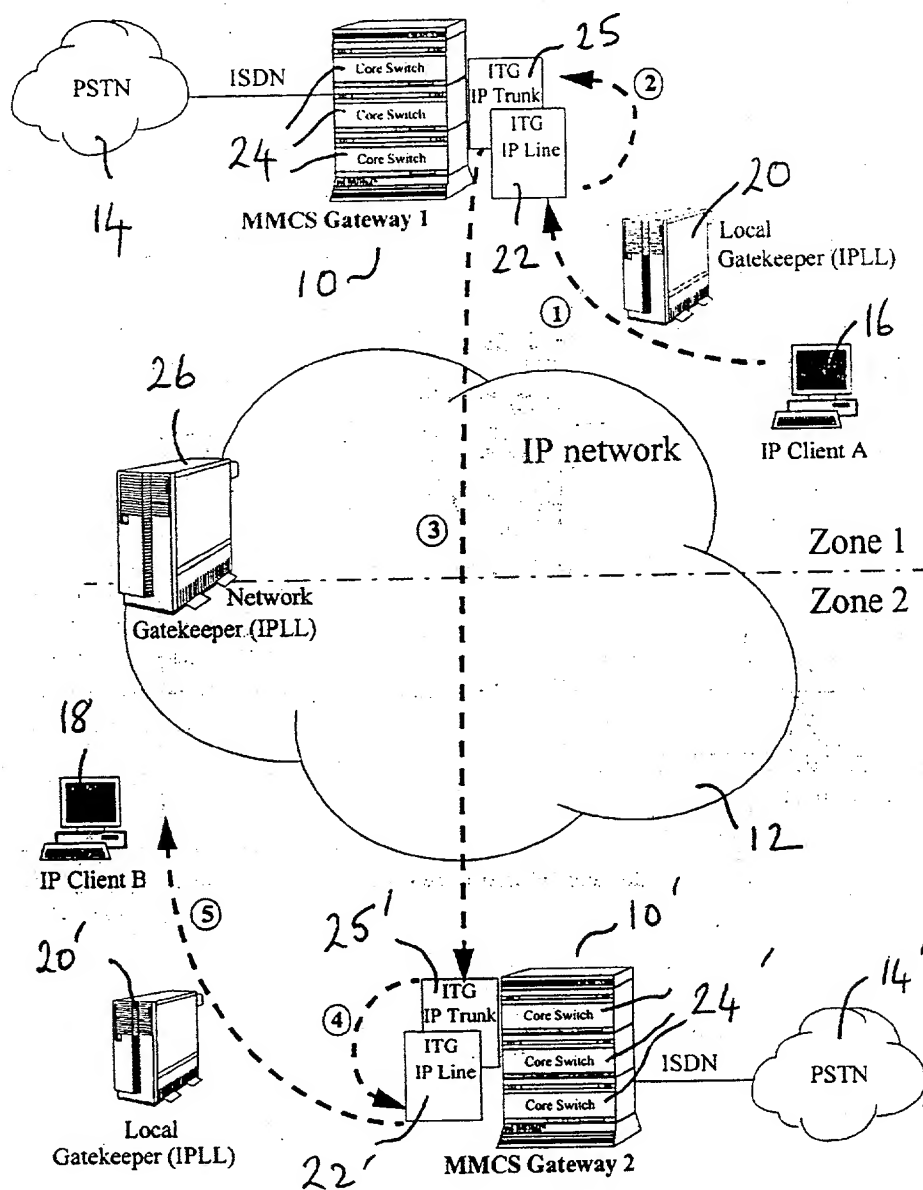


Fig. 6

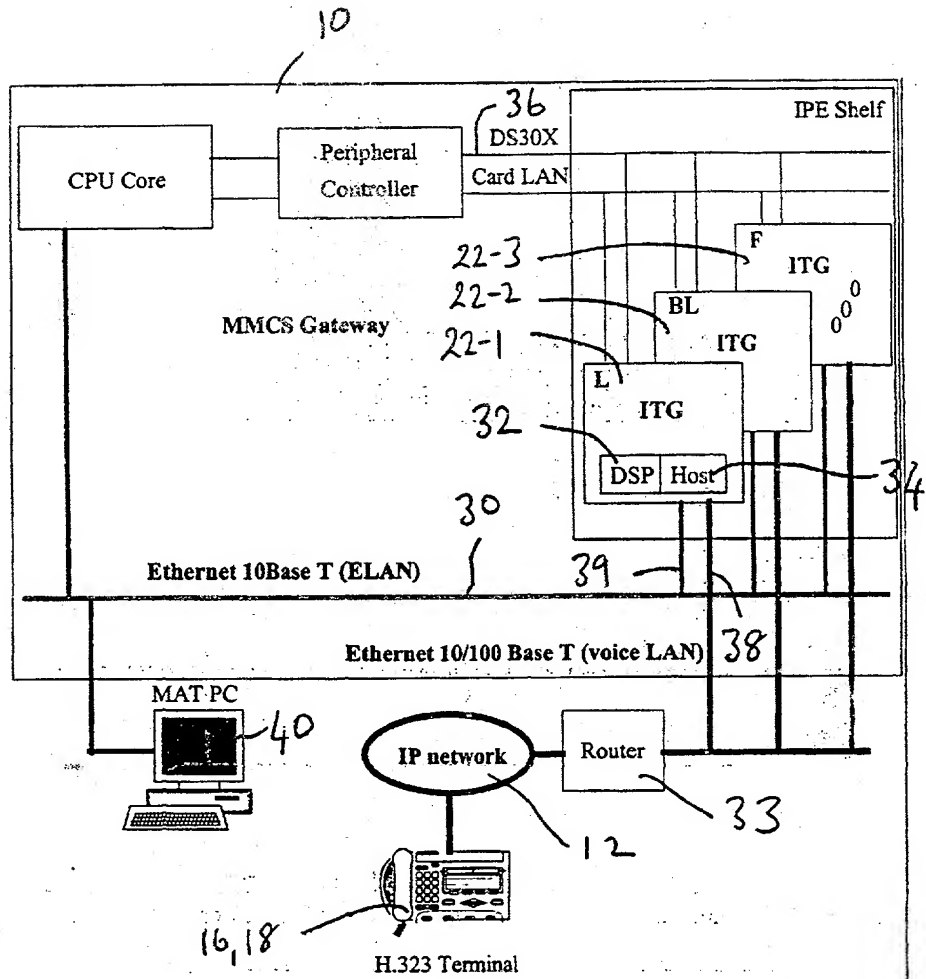


Fig. 7

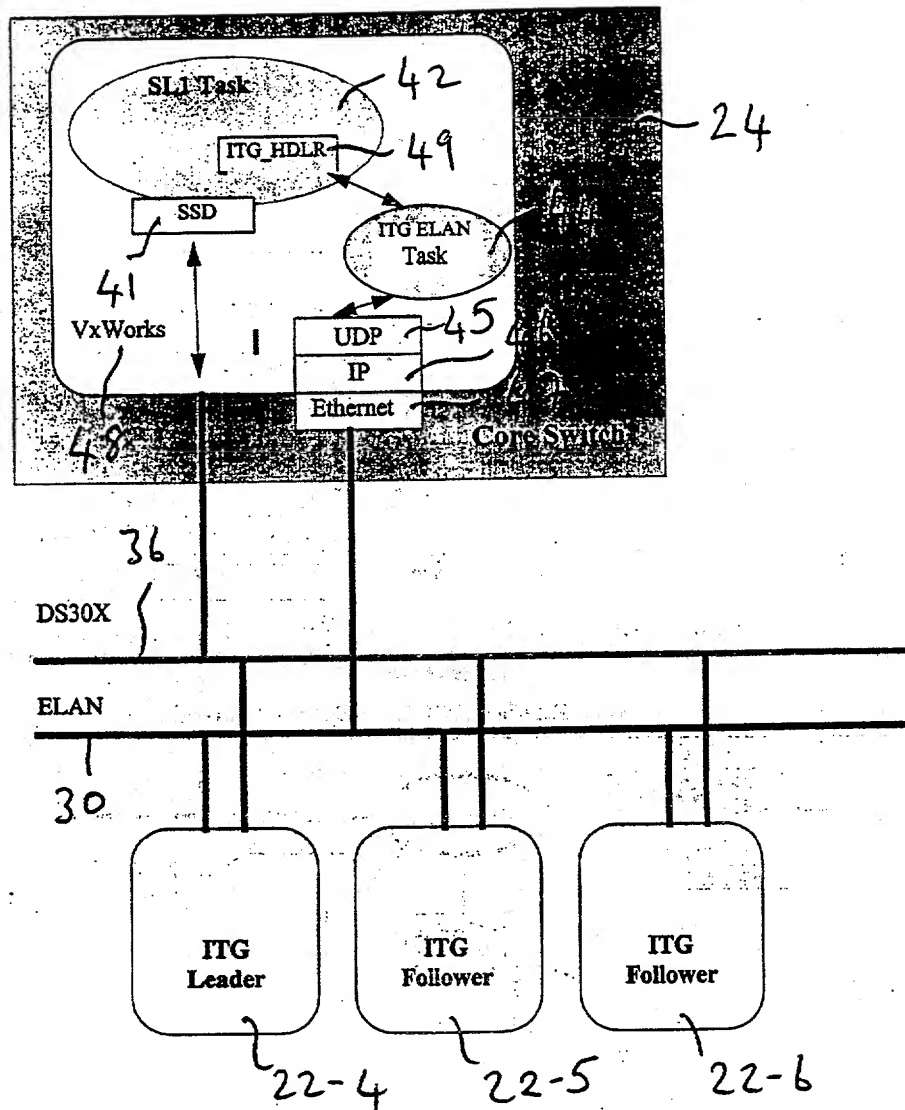


Fig. 8

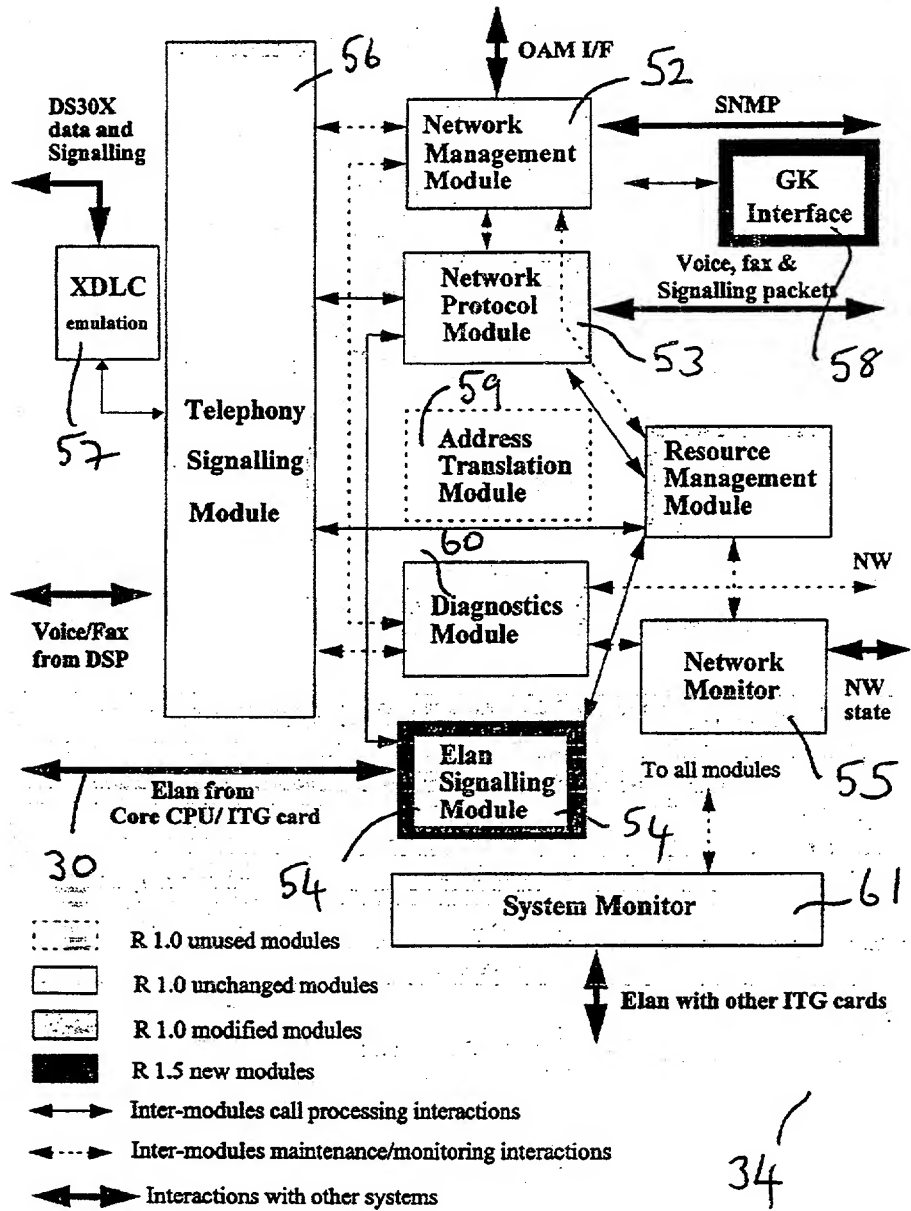


Fig. 9

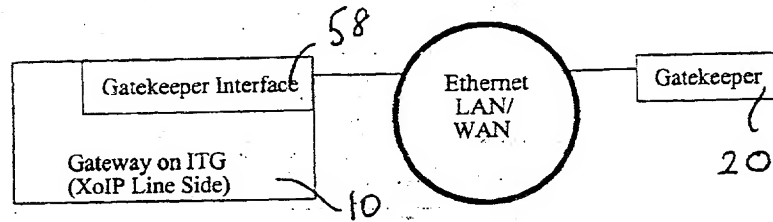
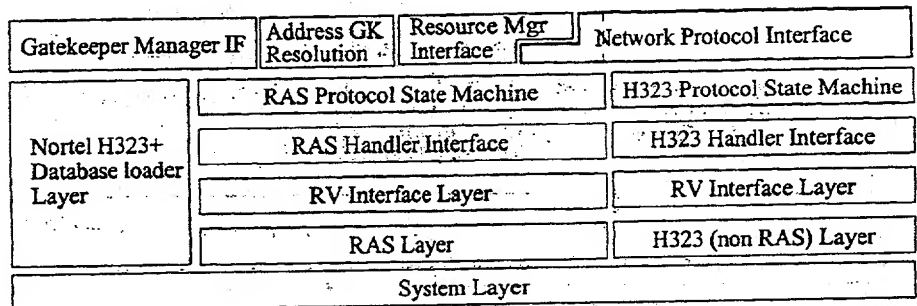


Fig. 10



☐ Gatekeeper specific layers
 ☐ RADVision Stack

Fig. 11

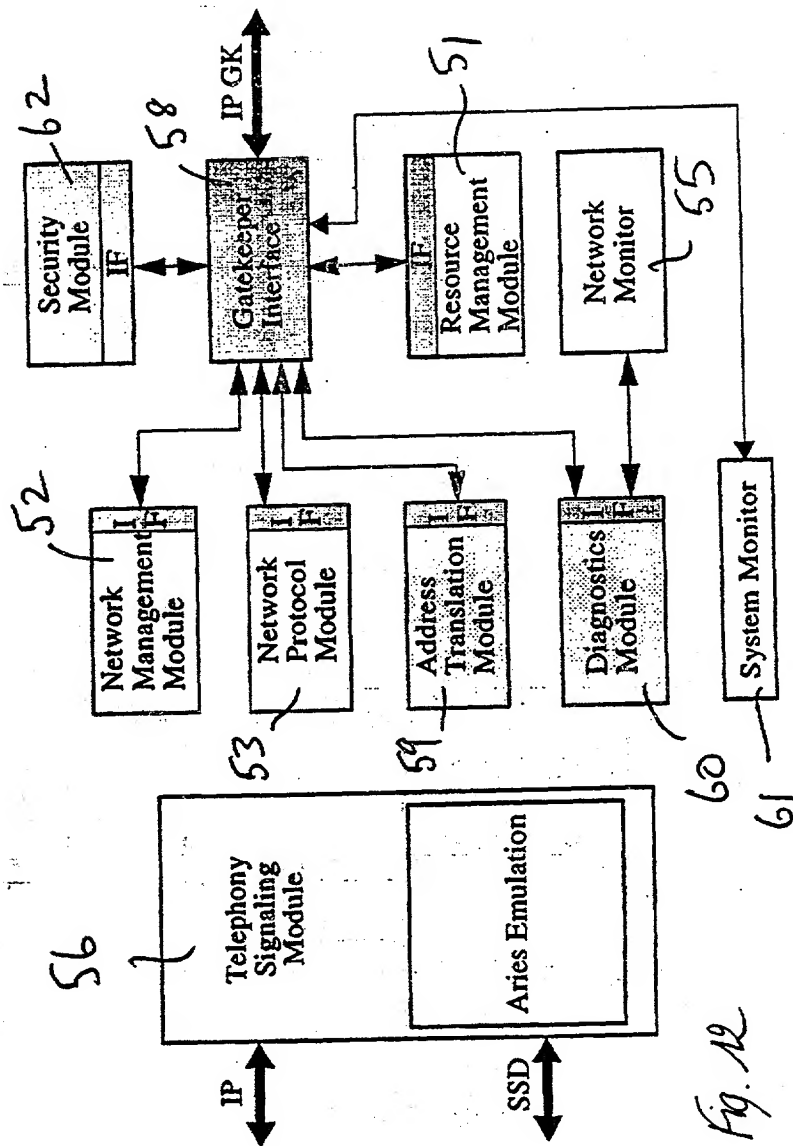


Fig. 12

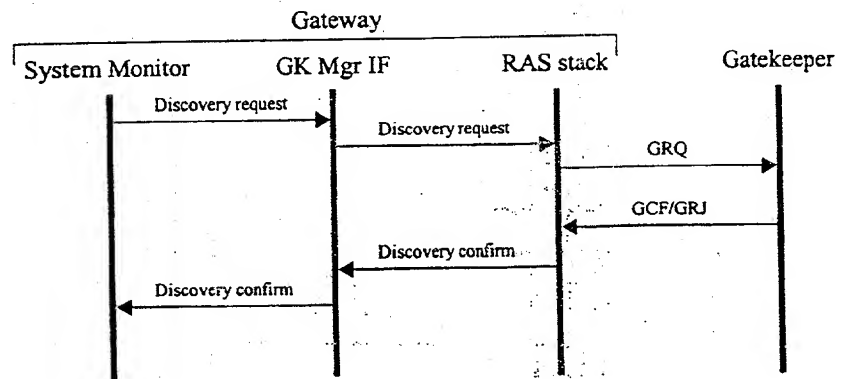


Fig. 13

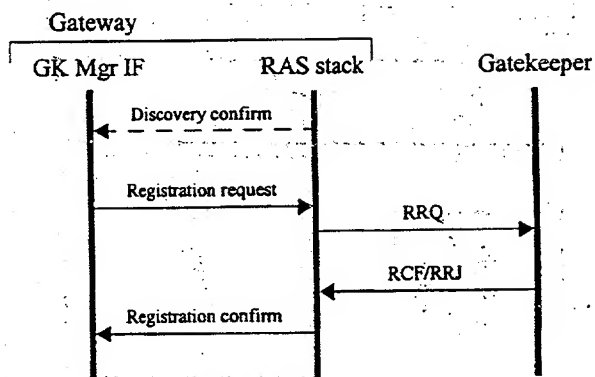
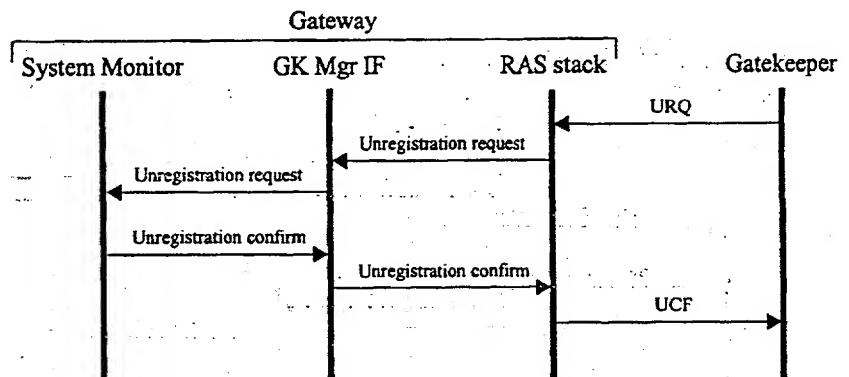
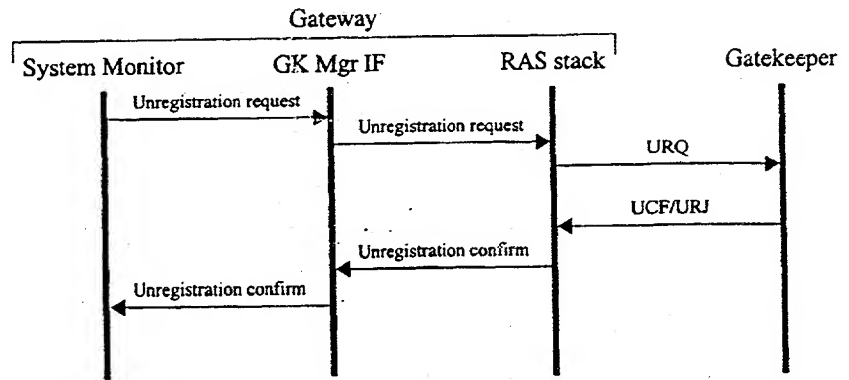


Fig. 14



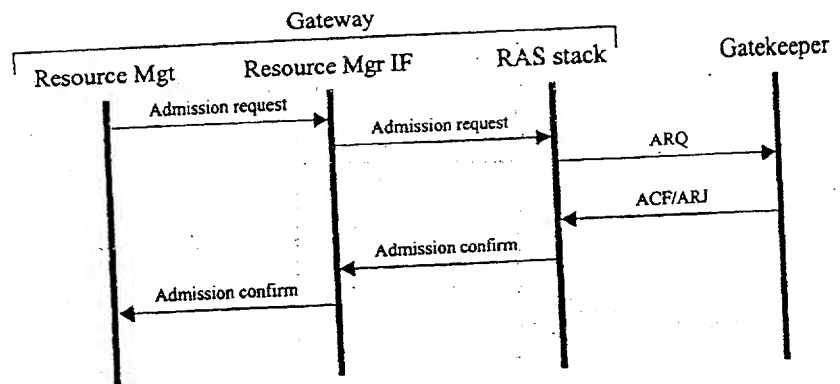


Fig. 17

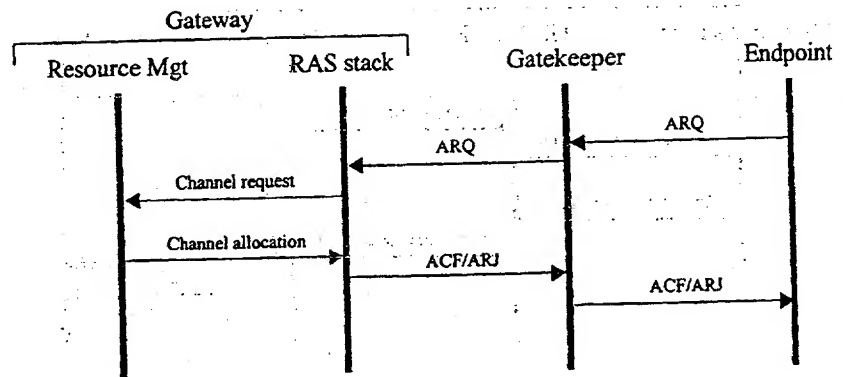


Fig. 18

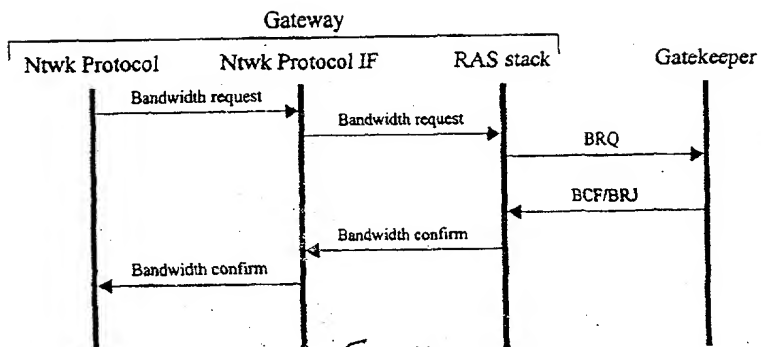


Fig. 19

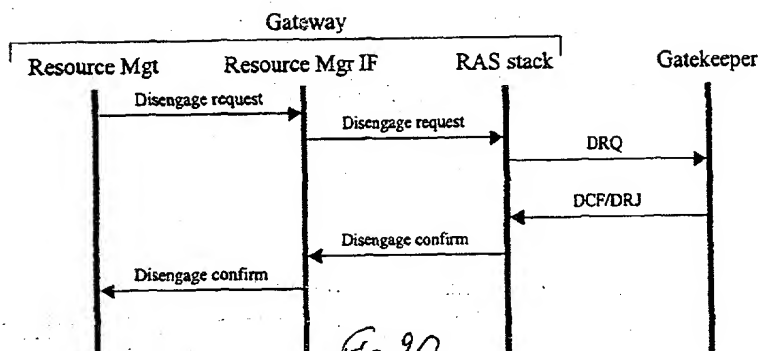


Fig. 20

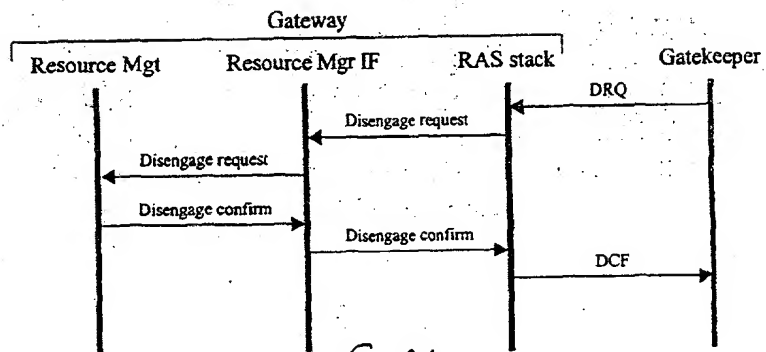
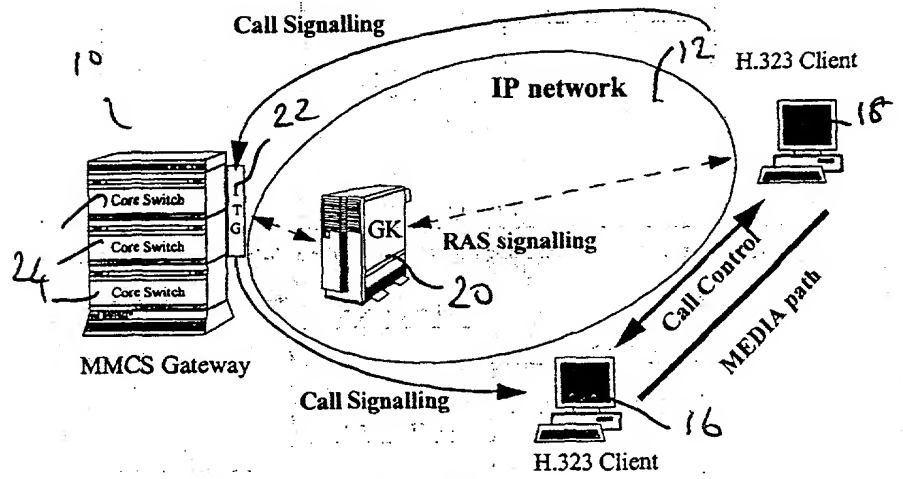
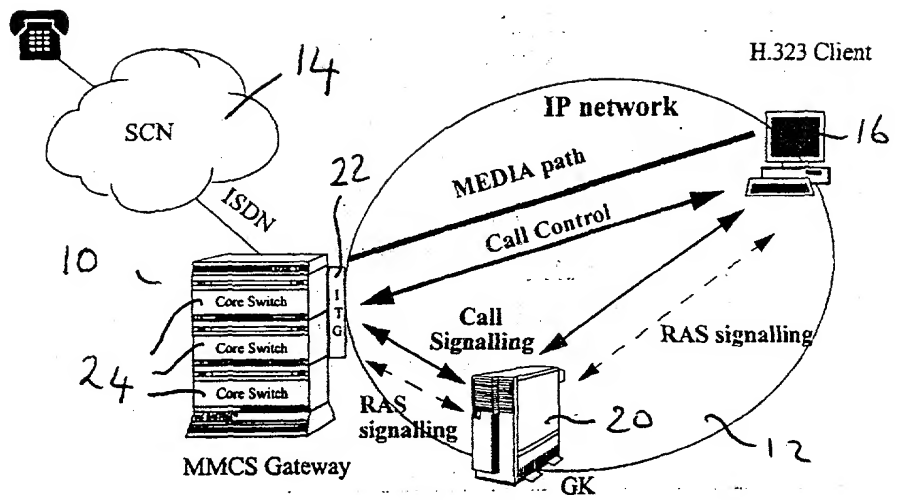


Fig. 21



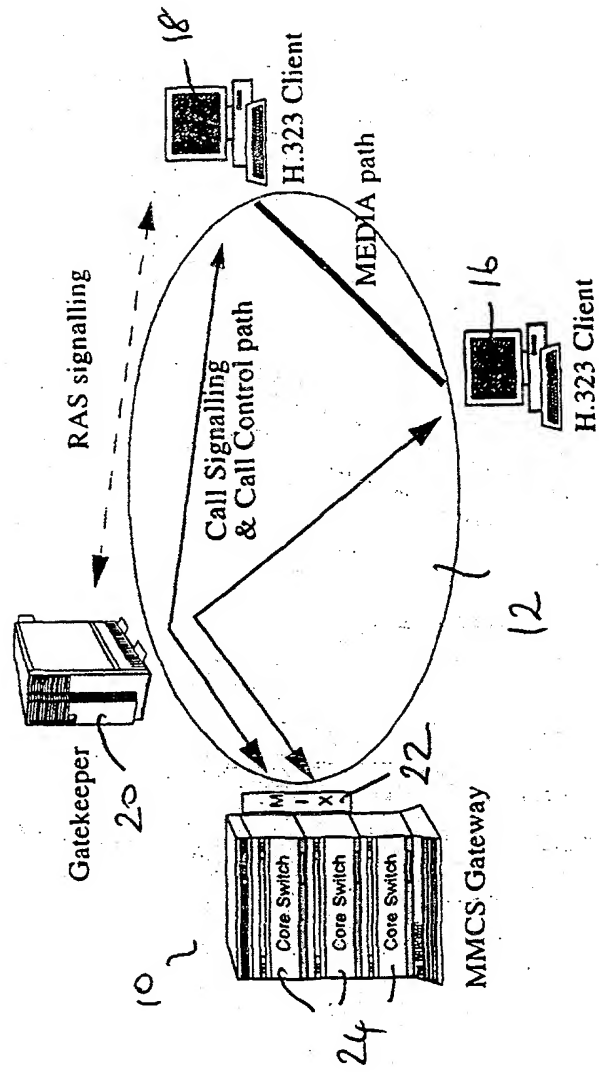


Fig. 23

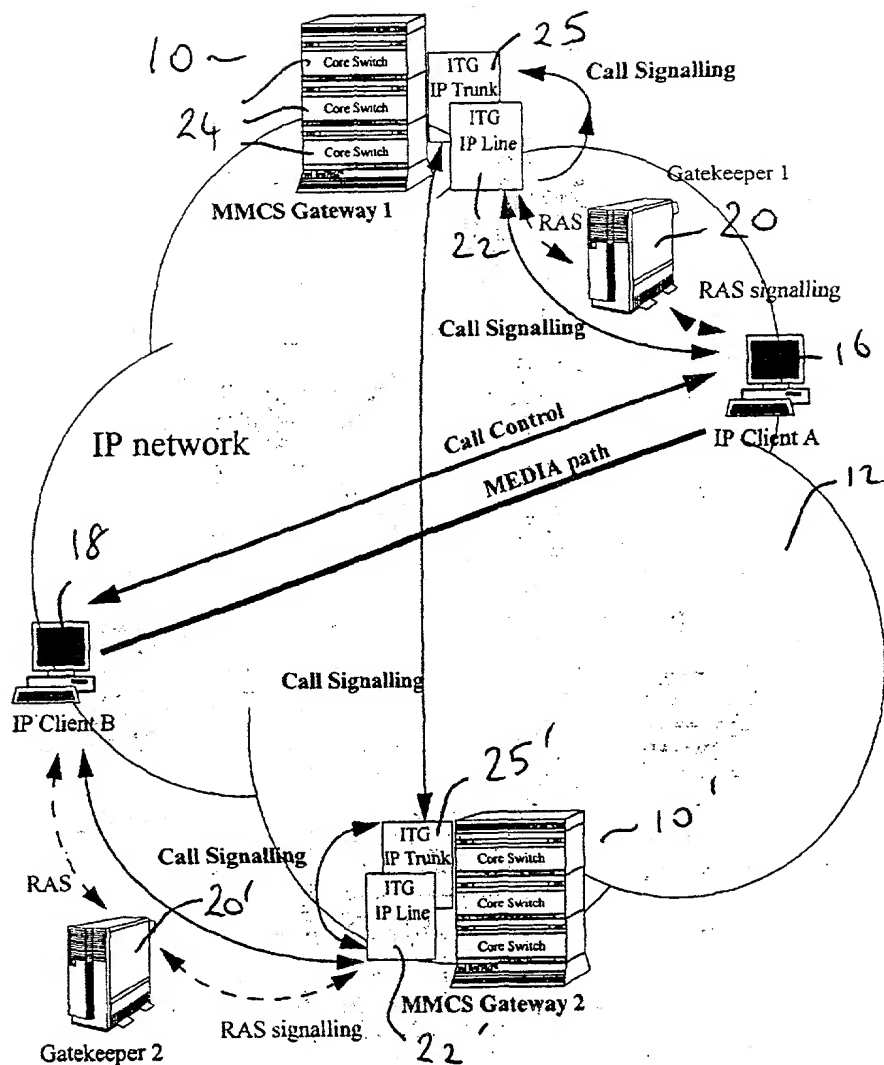


Fig. 25

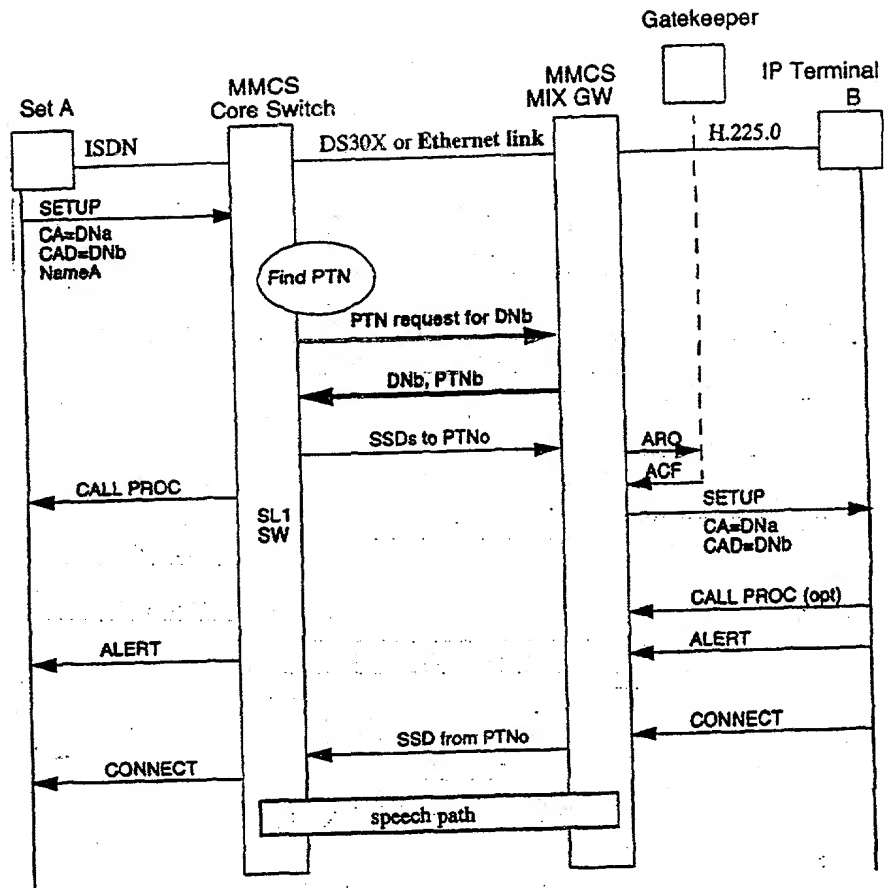


Fig. 26

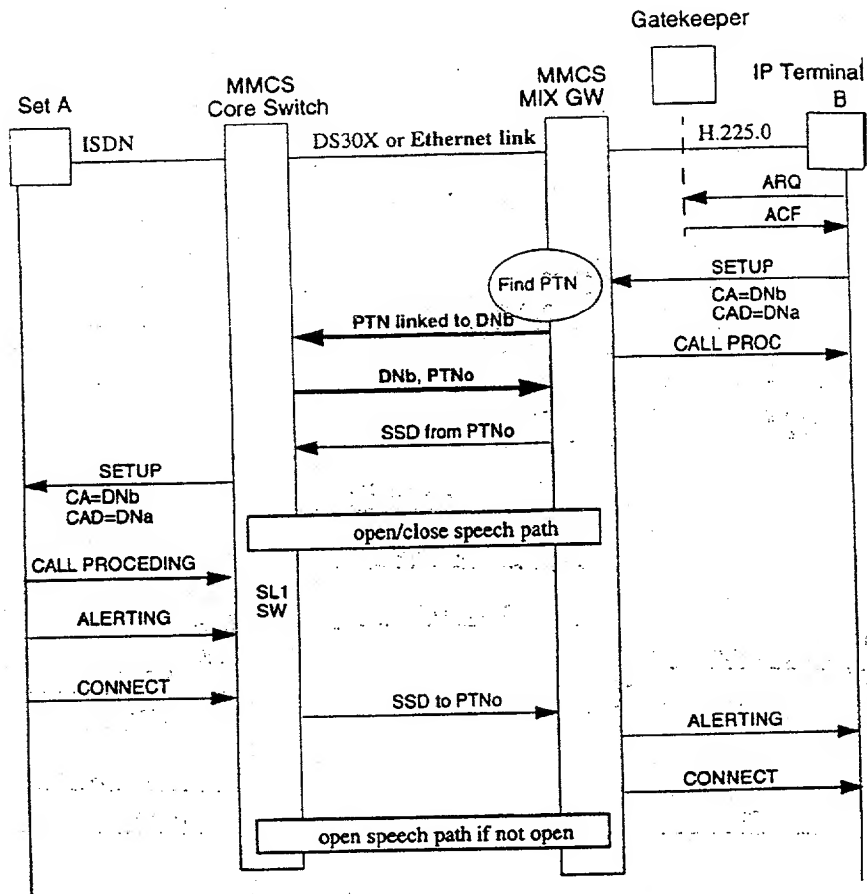


Fig. 27

- Q.931 messages (ISDN or H.225.0 call signalling)
- - - H.225.0 RAS signalling
- ELAN messages
- new SSD message
- existing SSD

Fig. 28

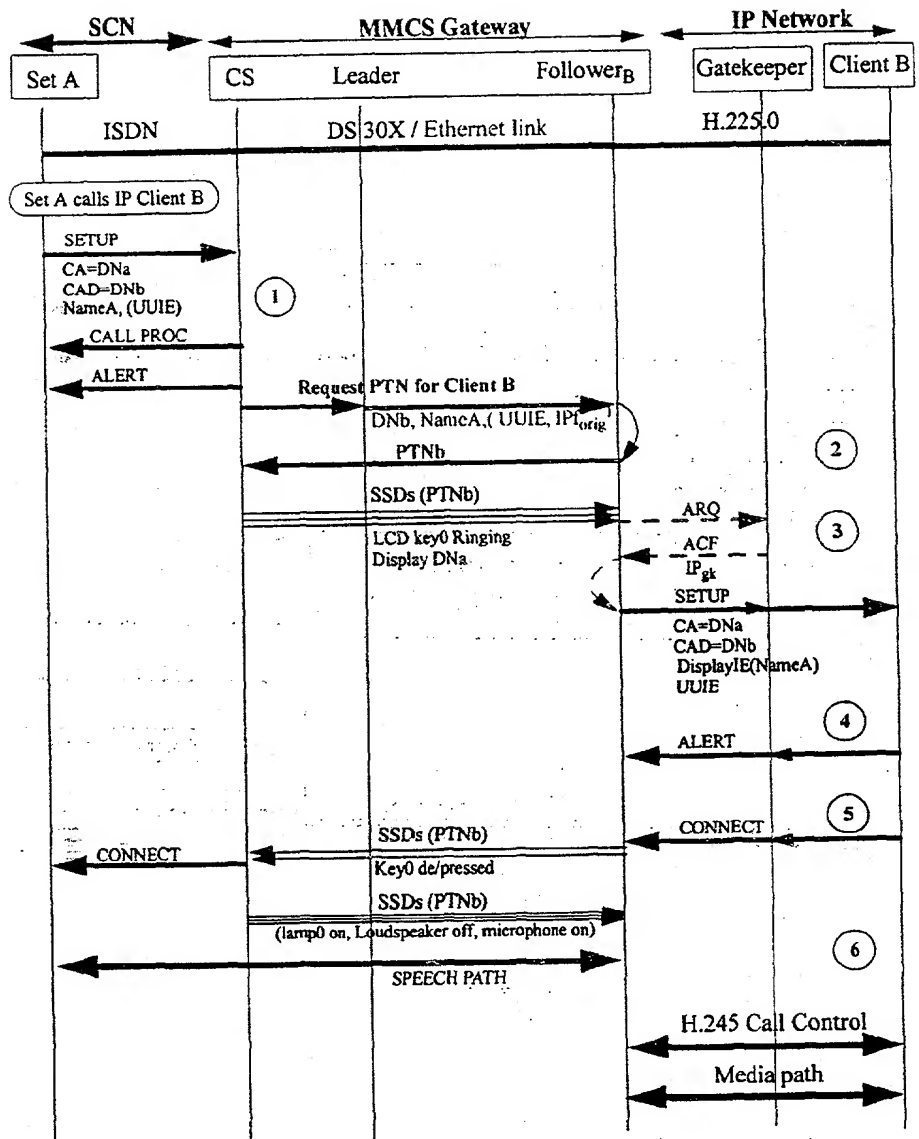


Fig. 29

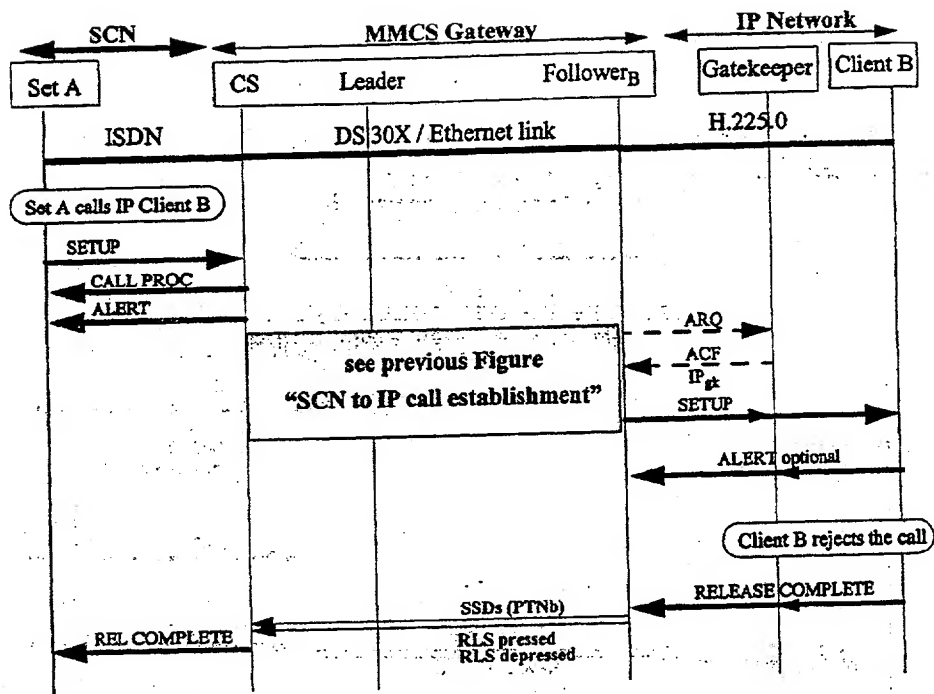


Fig. 30

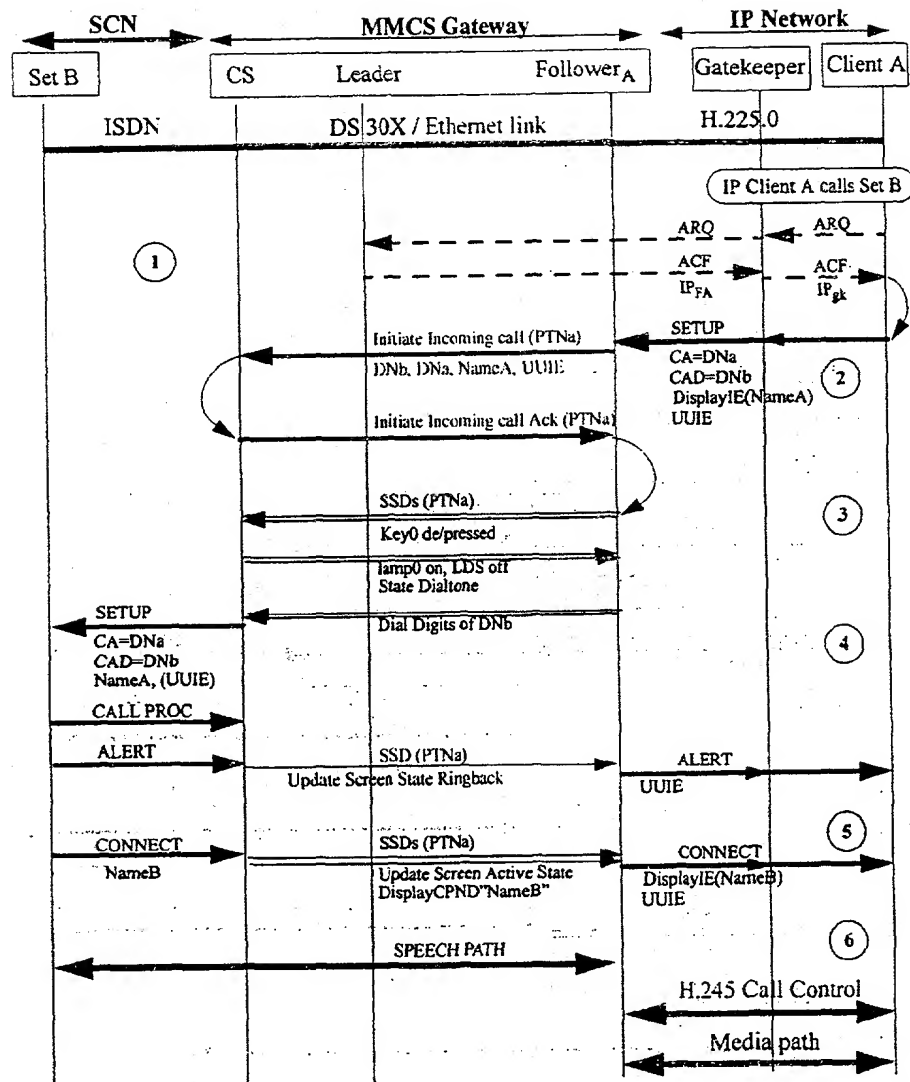
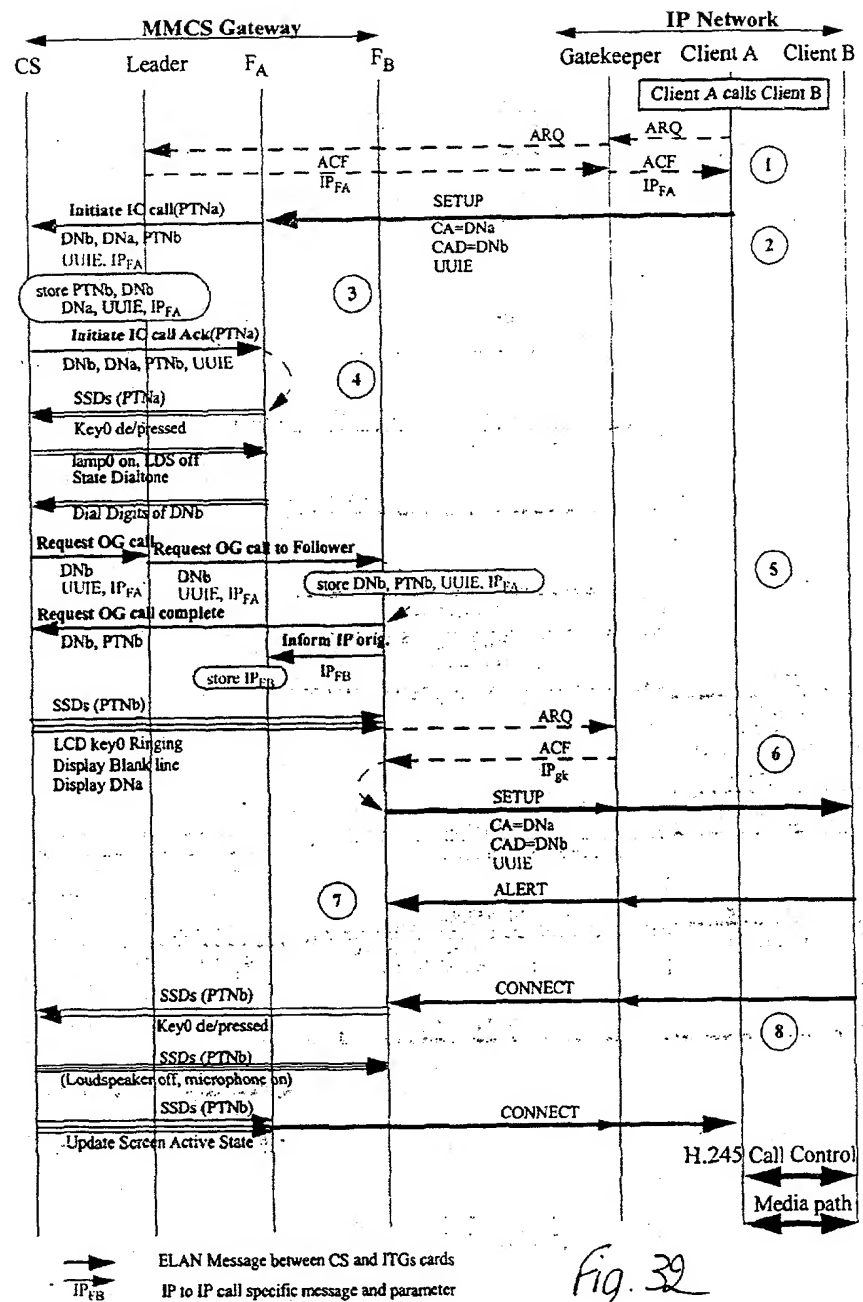


Fig. 31



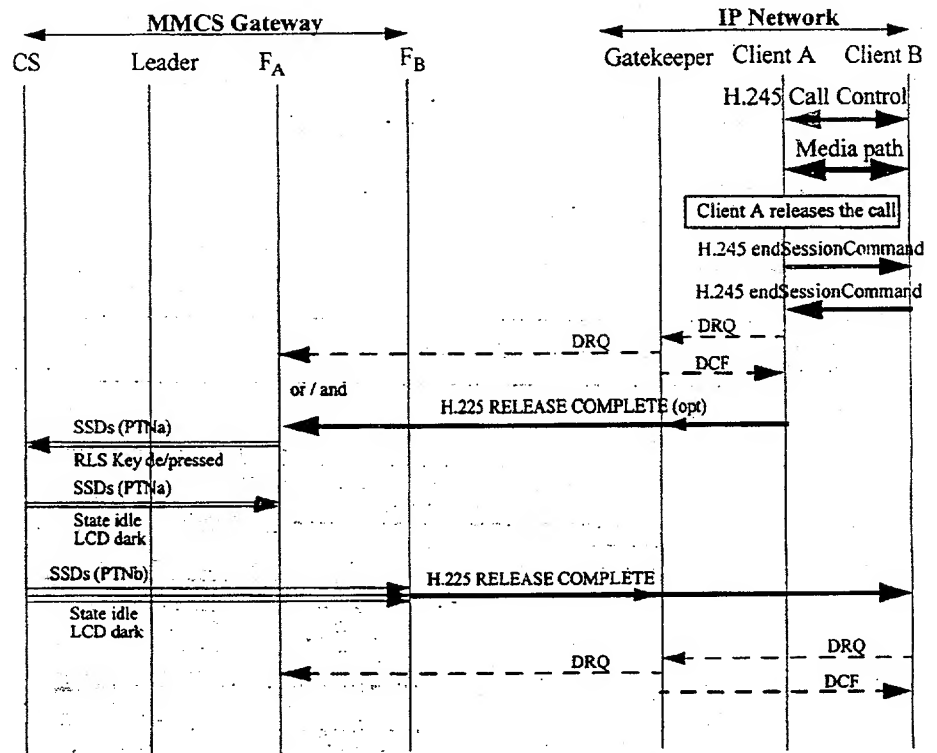


Fig. 33

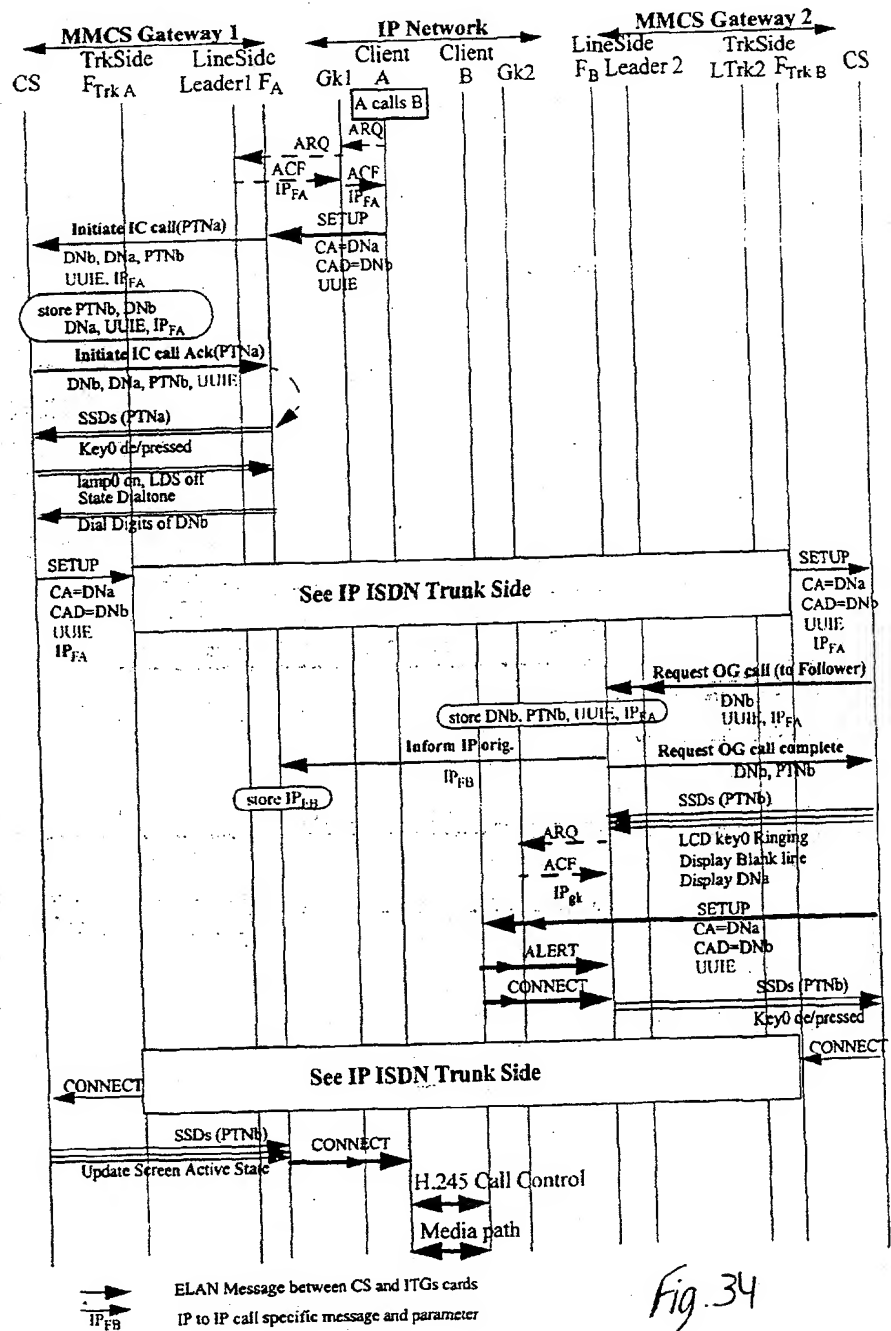


Fig. 34

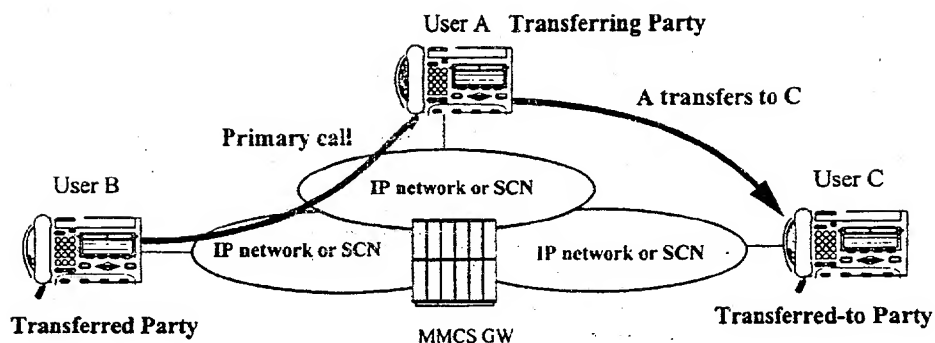


Fig. 35

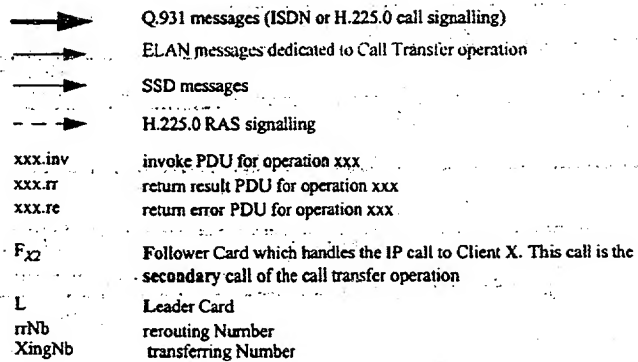


Fig. 36

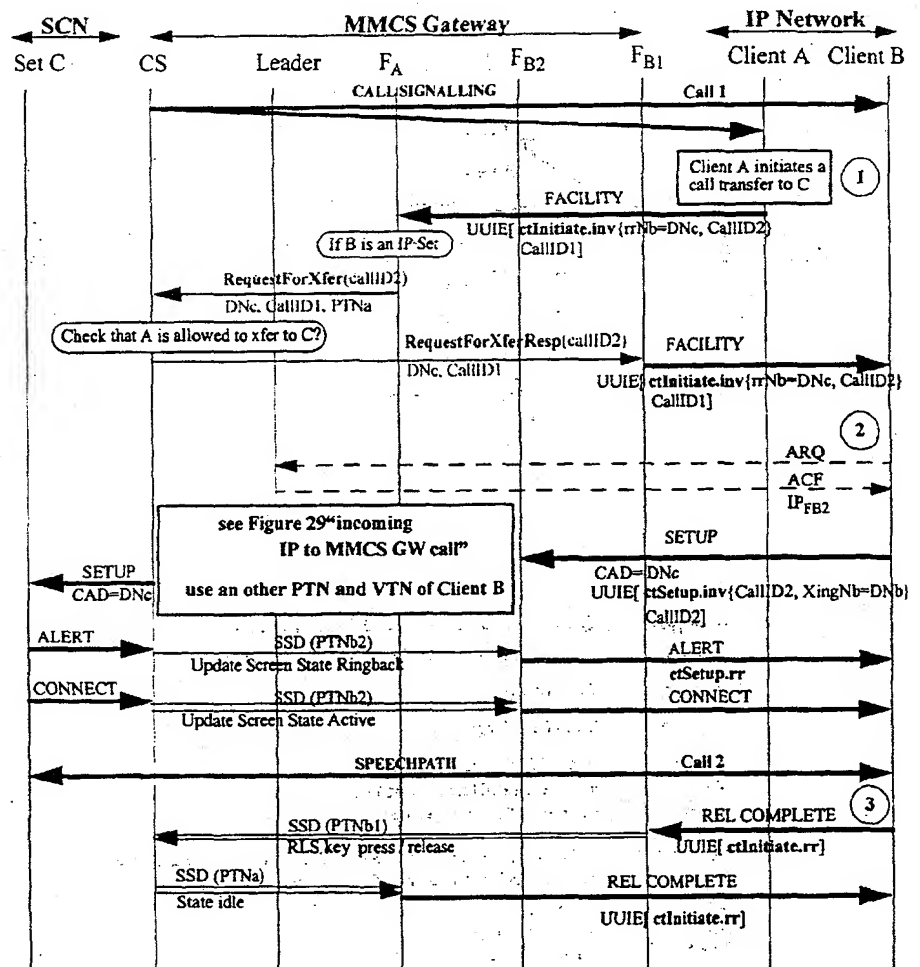


Fig. 37

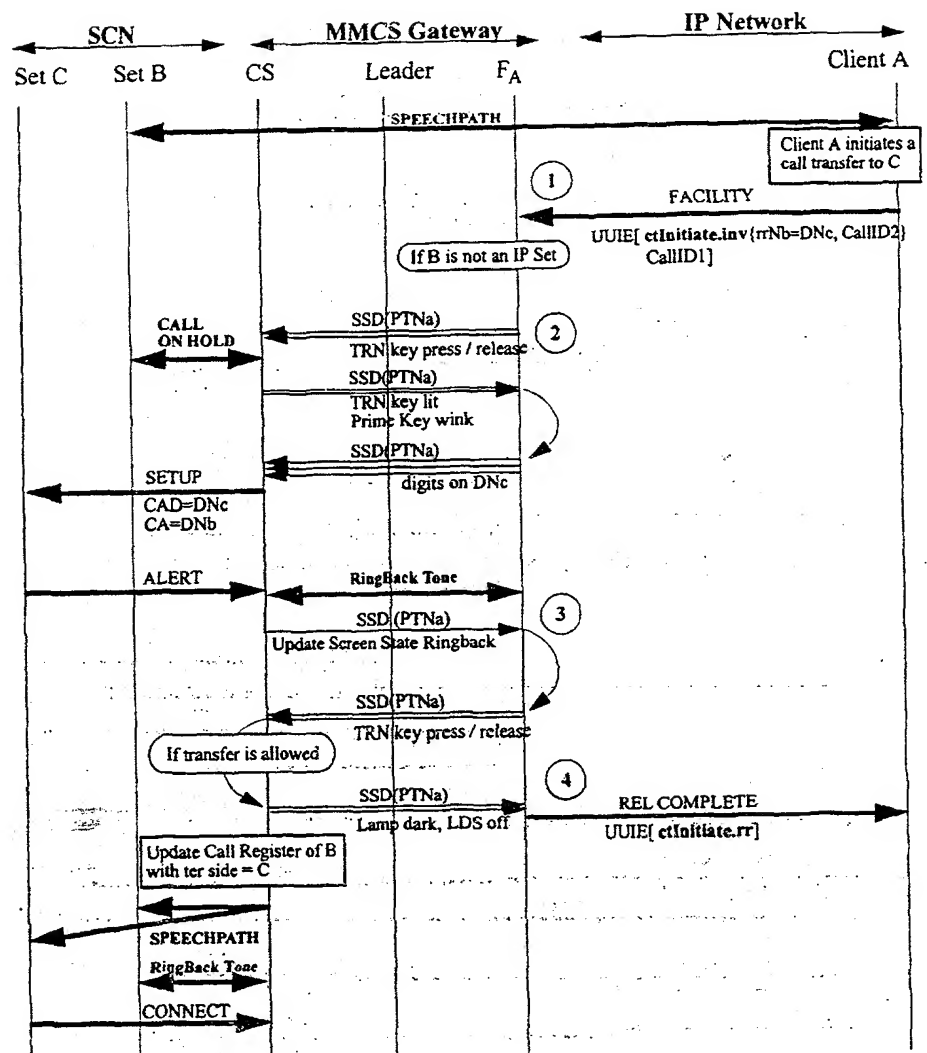


Fig. 38

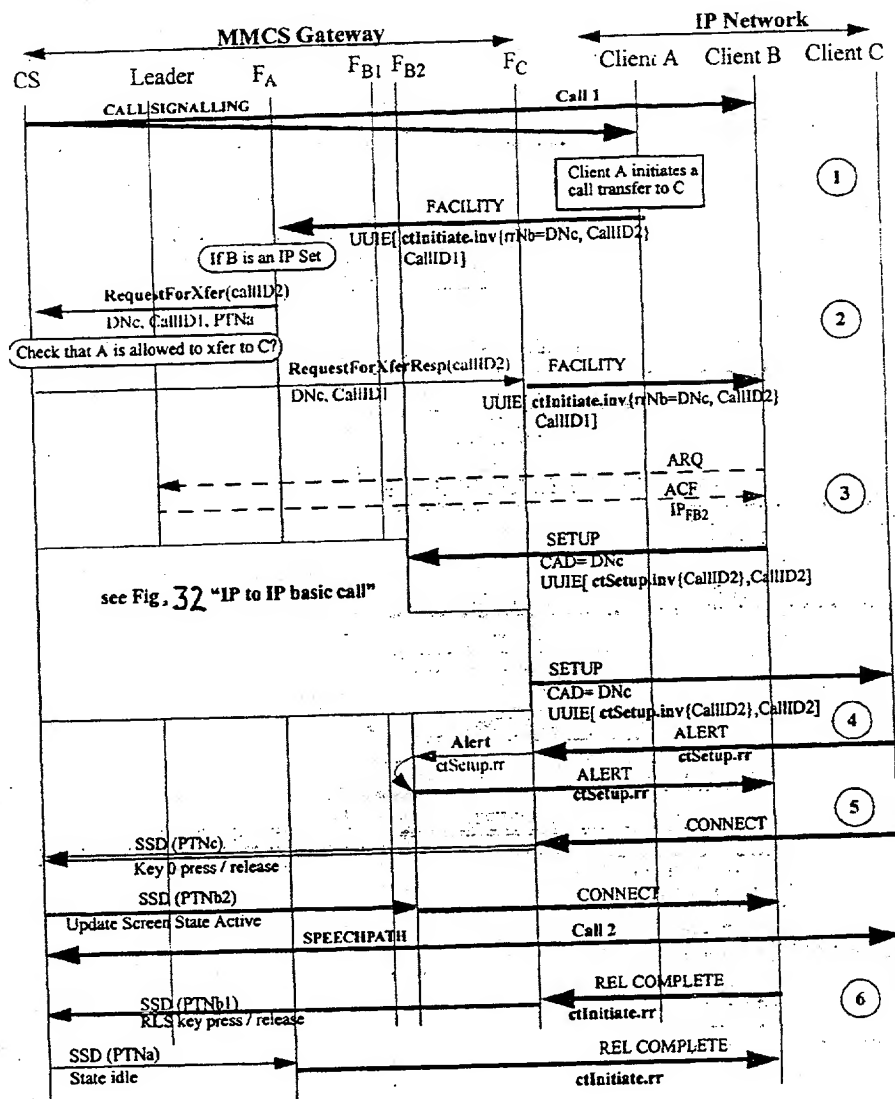


Fig. 39

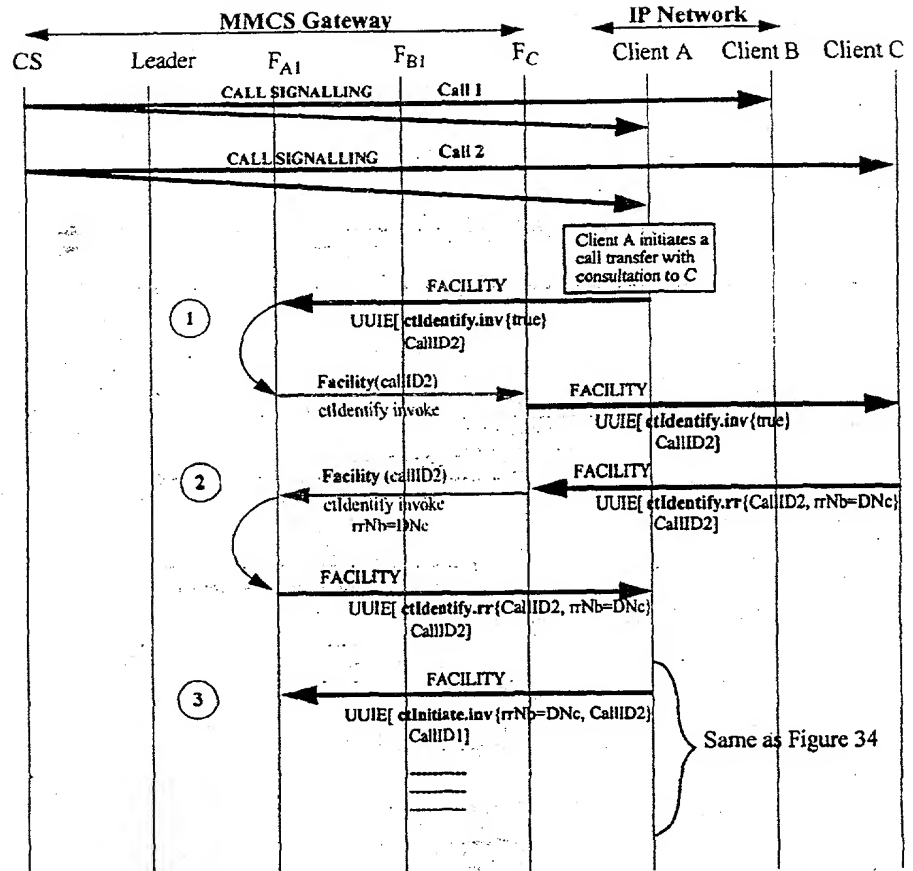


Fig. 40

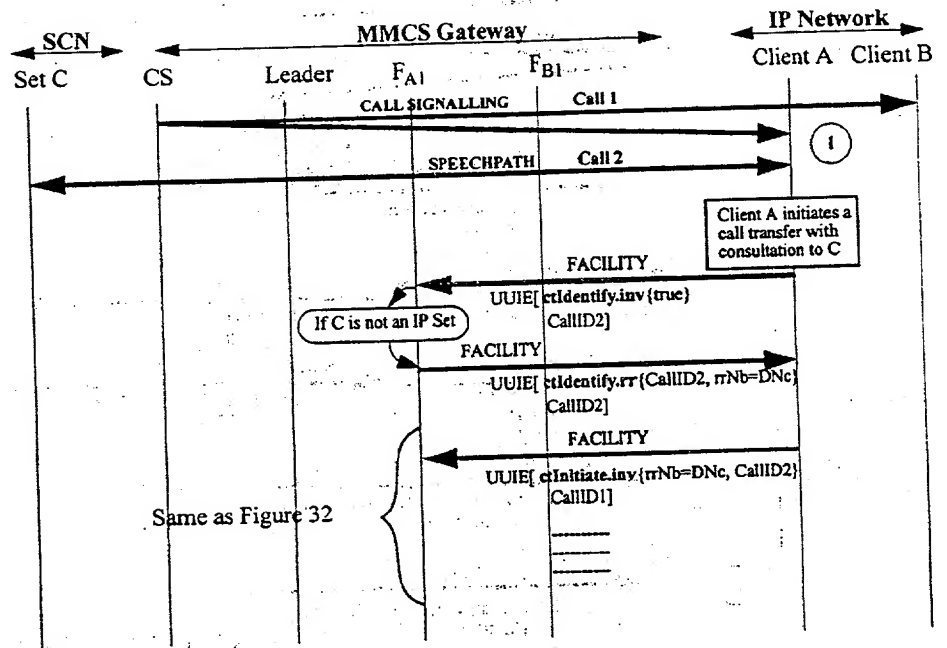


Fig. 41

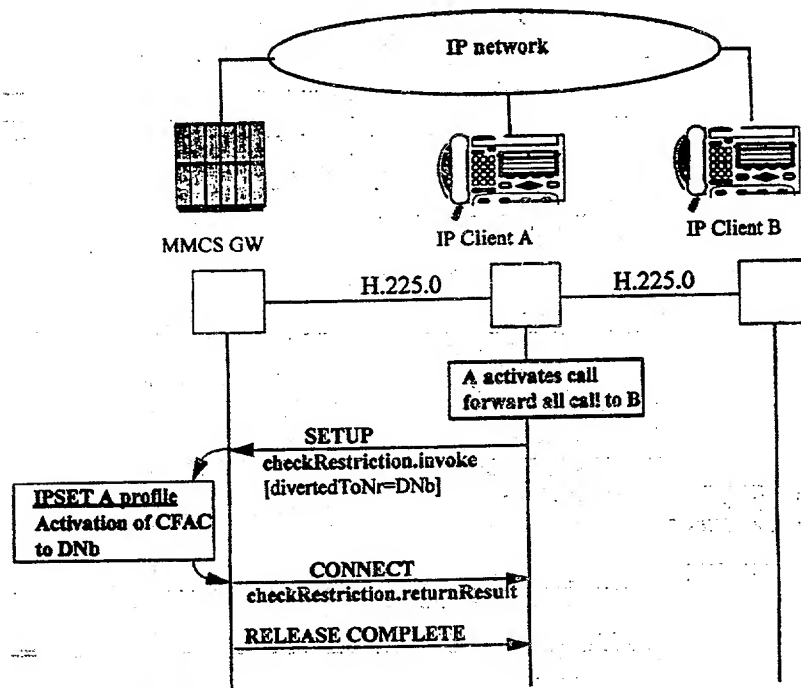


Fig. 42

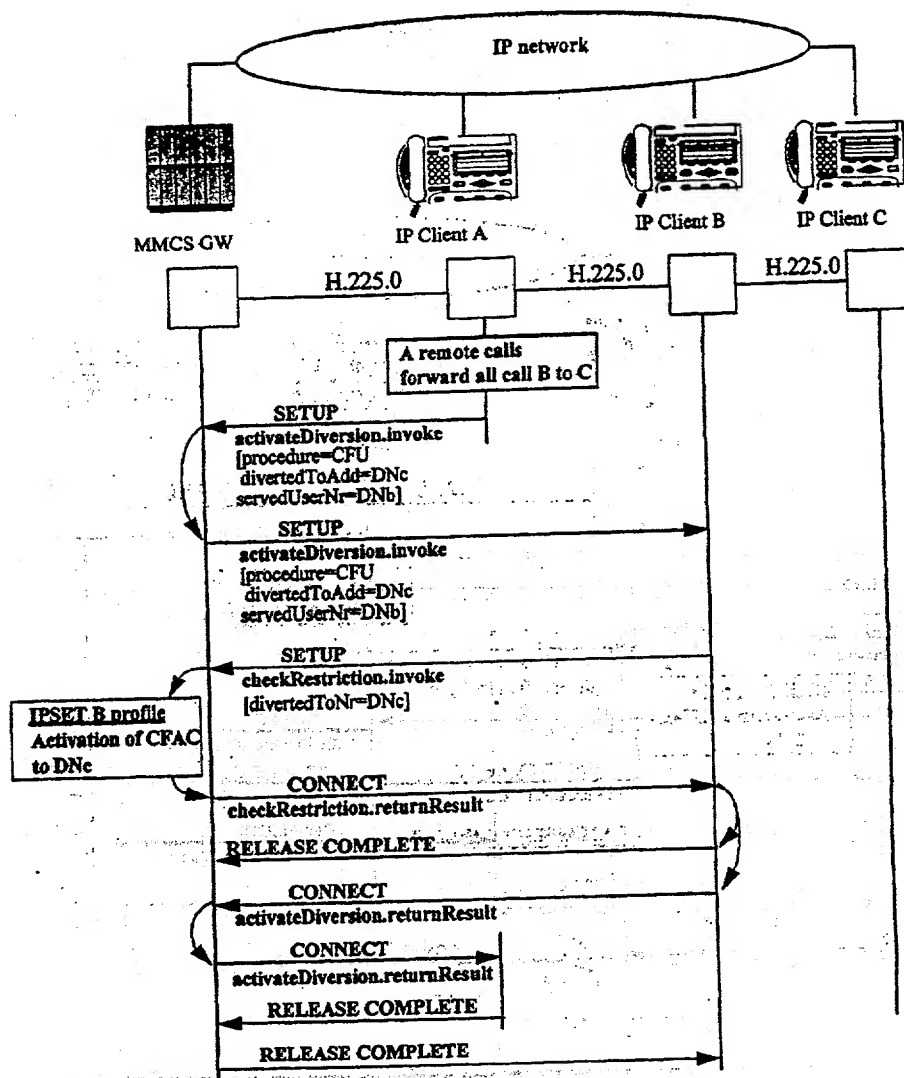
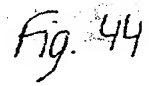


Fig. 43





(12) **CORRECTED EUROPEAN PATENT APPLICATION**

Note: Bibliography reflects the latest situation

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(71) Applicant: **Nortel Networks Limited**
Montreal, Quebec H2Y 3Y4 (CA)

(72) Inventors:
• **Brivet, Mayeul**
94350 Villiers sur Marne (FR)
• **Tripps, John**
75004 Paris (FR)
• **Aubry, Isabelle c/o Nortel Europe S.A.**
Bussy-St-Georges, 77608 Marne-la-Vallée (FR)

- **Colle, Pierre c/o Nortel Europe S.A.**
Bussy-St-Georges, 77608 Marne-la-Vallée (FR)
- **Fouletier, Jérôme c/o Nortel Europe S.A.**
Bussy-St-Georges, 77608 Marne-la-Vallée (FR)
- **Liotard, Michel c/o Nortel Europe S.A.**
Bussy-St-Georges, 77608 Marne-la-Vallée (FR)
- **Litzler, Serge c/o Nortel Europe S.A.**
Bussy-St-Georges, 77608 Marne-la-Vallée (FR)
- **Métairieau, Pierre-Yves c/o Nortel Europe S.A.**
Bussy-St-Georges, 77608 Marne-la-Vallée (FR)

(74) Representative: **Bewley, Ewan Stuart et al**
Nortel Networks
Intellectual Property Law Group
London Road
Harlow, Essex CM17 9NA (GB)

(54) **IP telephony gateway**

(57) The present invention provides an IP telephony gateway. According to a first aspect of the invention, the gateway provides communications between a switched circuit network (SCN) and an IP network. The gateway can handle calls between clients on the switched circuit network and IP clients on the IP network. The gateway provides supplementary call services/features for calls to/from IP clients on the IP network, thus providing IP clients with similar features to those that are available to terminals on a PBX. The gateway is preferably a PBX which supports the supplementary services/features.

Advantageously, the gateway can also provide sup-

plementary call services/features to calls between IP clients on the IP network. This can be achieved by routing call control signaling for IP client - IP client calls via the gateway where the services can be controlled.

A further aspect of the invention provides an IP network in which IP clients have access to a range of supplementary call features/services. At least one of the supplementary features/services is provided by a gateway, such as a PBX, at an interface to the IP network. A call from an IP client is routed via the gateway to apply the supplementary feature/service.

EP 0 966 145 A8

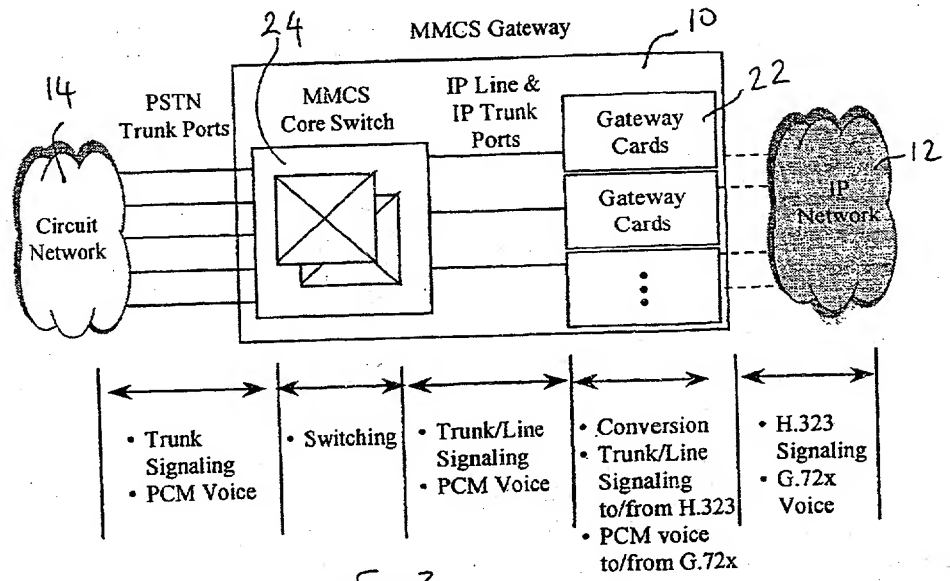


Fig. 3